TAPR HF Special Interest Group Archive, January 1997 through December 1998

January 1997 through December 1998

TAPR's forum for experimentation and development of HF digital communications, including modulation, coding, protocols, DSP platforms, modems, propagation, and on-air testing.

This volume contains 24 original monthly files and 1,120 messages, arranged chronologically. The PDF is a readable access copy: MIME transfer encodings are decoded, HTML bodies are converted to plain text, and non-text attachments are represented by descriptive notes. The raw source text is supplied separately with the Internet Archive item.

Month	Messages	Original source file
1997-01	41	hfsig.9701
1997-02	90	hfsig.9702
1997-03	27	hfsig.9703
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1997-05	36	hfsig.9705
1997-06	89	hfsig.9706
1997-07	170	hfsig.9707
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1997-09	54	hfsig.9709
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1998-01	49	hfsig.9801
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1998-03	10	hfsig.9803
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1998-07	12	hfsig.9807
1998-08	66	hfsig.9808
1998-09	50	hfsig.9809
1998-10	85	hfsig.9810
1998-11	47	hfsig.9811
1998-12	43	hfsig.9812

January 1997

Original source file: hfsig.9701
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9701

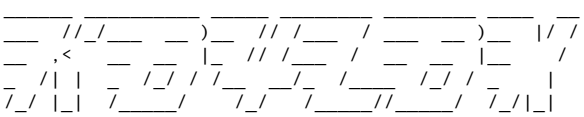
1. pactor

From: Alex Evonosky <alexevon@abraxis.com>
Date: Wed, 01 Jan 1997 17:25:09 -0500
To: hfsig@tapr.org
Cc: kb4lbx@amsat.org
Message-ID: <1.5.4.32.19970101222509.0068b2d8@smtp1.abraxis.com>
Rendered body type: text/plain

I recently purchased the PacCom PTC-II. I am wondering where do most of the
Pactor sig run around? Wanted to test this unit out and see how it copies
compared to the AEA-DSP232.
Thank You in Advance.

Best 73,
de KB4LBX/Alex

Alex F. Evonosky, KB4LBX   OSCAR P3D U/V,V/U,L/U
Alpharetta, GA / EM74   Packet LAN 145.07MHz
WAS WAC ARRL AMSAT   ATV 1.2GHz/421.25MHz
kb4lbx@amsat.org   160m-1.2GHz SSB/CW/Digital/SSTV
                           GRAPES 56kB Packet WAN
                           Active HF/VHF contester
                           DX4WIN Computer Logging System
                           W95SSTV SSTV software



2. Re: [HFSIG:1871] Re: Parallel Serial tone modem comparison

From: mcdermot@rdxsunhost.aud.alcatel.com (Tom Mcdermott)
Date: Thu, 2 Jan 97 07:52:58 CST
To: hfsig@tapr.org
Message-ID: <9701021352.AA02325@eagle.aud.alcatel.com>
Rendered body type: text/plain

> It is hf95_6.html
>
> You should be able to find it going through the menus on that page. It is
> definately there though. It also talks about the difference between AWGN and
> real measured noise (including lightning) on these modems.
>
>
> - Charles
>
>

        Thanks, Charles. I found it and printed it. A good paper,
and it certainly paints positive hints towards the value of
trellis coded modulation (TCM). Will have to wait to next week to
pull the references on trellis codes for the Rayleigh channel.
Thanks for the pointer.

                                                - Tom

Tom McDermott
mcdermot@aud.alcatel.com

3. Re: [HFSIG:1895] Re: Parallel Serial tone modem comparison

From: Charles Brain <chbrain@dircon.co.uk>
Date: Thu, 02 Jan 1997 16:41:43 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970102164143.00678b0c@popmail.dircon.co.uk>
Rendered body type: text/plain

At 08:00 02/01/97 -0600, you wrote:
>
>> It is hf95_6.html
>>

>> You should be able to find it going through the menus on that page. It is
>> definately there though. It also talks about the difference between AWGN and
>> real measured noise (including lightning) on these modems.
>>
>>
>> - Charles
>>
>>
>
>
>   Thanks, Charles. I found it and printed it. A good paper,
>and it certainly paints positive hints towards the value of
>trellis coded modulation (TCM). Will have to wait to next week to
>pull the references on trellis codes for the Rayleigh channel.
>Thanks for the pointer.
>
>   - Tom
>
>
>Tom McDermott
>mcdermot@aud.alcatel.com
>
>
>
I am glad you found it of interest. I have a copy of a paper written by the
same author about the DSTO modem that uses Ungerboeck coding. Unfortunately
I only have it in paper not electronic form it was from the IEE Conference
on HF Radio Systems and Techniques 1994.

Regards Charles

4. Re: [HFSIG:1892] Re: 10m FM packet

From: Erik Skovgaard <Erik_Skovgaard@bc.sympatico.ca>
Date: Sat, 04 Jan 1997 21:53:13 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970105055313.00690728@mail.bctel.ca>
Rendered body type: text/plain

Sounds like a specific US issue. In Canada we are allowed to operate any
mode on 10 meters as long as we stay within a 20 kHz limit.

OTOH - why waste a good band....

73 de VE7MDL   ....Erik.

At 03:13 30/12/96 -0600, you wrote:
>At 02:16 30/12/96 -0600, you wrote:
>>Anyone lend an ear or thought to allow 1200 AFSK in FM portion of 10m
>>band
>>(lower FM band edge to repeaters) and permit digi ?
>>73's
>>andy/kf4xh@juno.com
>>
>>
>>
>Yes this is something I believe TAPR should press for. The BW limit on
>data modes at H.F is seriously hampering new developments. Althought my
>U.K licence doesn't impose such a limit if U.S amateurs arn't allowed
>to use wider bandwidths then any new development on these lines is a non
>starter. How voice bandwith modems are going to be accomodated I don't
>know. They won't be welcome in the RTTY subsections and just look at the
>abuse Slow Scanners have to contend with in the voice sections!
>
>It maybe necessary to only use them on specially licenced point to point
>links to augment the BBS traffic carried by satellite.
>
>Also on a different subject. It would be nice to promote a decent waveform
>to be used on the HF beacon network so we can automatically gather SINAD and
>MP information. The current system is pretty useless.
>
>- Charles (G4GUO)
>
>
>

5. Re: [HFSIG:1897] Re: 10m FM packet

From: Charles Brain <chbrain@dircon.co.uk>
Date: Sun, 05 Jan 1997 09:34:56 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970105093456.00678368@popmail.dircon.co.uk>
Rendered body type: text/plain

At 23:55 04/01/97 -0600, you wrote:
>Sounds like a specific US issue. In Canada we are allowed to operate any
>mode on 10 meters as long as we stay within a 20 kHz limit.
>
>OTOH - why waste a good band....
>
>73 de VE7MDL   ....Erik.
>
>
>At 03:13 30/12/96 -0600, you wrote:
>>At 02:16 30/12/96 -0600, you wrote:
>>>Anyone lend an ear or thought to allow 1200 AFSK in FM portion of 10m
>>>band
>>>(lower FM band edge to repeaters) and permit digi ?
>>>73's
>>>andy/kf4xh@juno.com
>>>
>>>
>>>
>>Yes this is something I believe TAPR should press for. The BW limit on

I am afraid I didn't look at the title of the message before I wrote HI.
I was referring to the lower bands. On 10m I don't think it is a problem,
but I am not familiar with the U.S regs. However comments have been made
in the past on the Sig about the BW limit on the lower bands.

By the way was there an HF blackout yesterday (SAT) or has my transceiver
gone wrong ?

- Charles

6. subscribe hfsig

From: "Frank Knoll" <fknoll@ibm.net>
Date: Sun, 05 Jan 97 15:21:48 +0100
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <199701051919.TAA36732@pop01.ny.us.ibm.net>
Rendered body type: text/plain

[Empty message body]

7. Re: [HFSIG:1884] Stanag 4529

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Mon, 06 Jan 1997 09:10:18 +0100
To: hfsig@tapr.org
Message-ID: <32D0B36A.144B8167@zeus.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

Charles Brain wrote:
>
> If anybody wants a copy of this STANAG (It is the narrow bandwidth version
> of 4285 I have found an electronic copy and can send it to them. It is a 70k
> zipped file that consists of a Word document and a powerpoint one.
>
> I don't think it is a good idea to post it directly to the sig and I can't
> remember where I found it so I can't point you there!!
>
> - Charles

Hi Charles!

Could you send me a copy?

73' Alexander

--
------------------------------+---------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| u. Elektroakustik | Fax : +49-6151-165545

```
|
| Merckstrasse 25               | EMail: a.kurpiers@uet.th-darmstadt.de
|
| D-64283 Darmstadt (Germany)   | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
*-------------------------------+--------------------------------------*
```

8. Re: [DSP:154] DSP files

```
From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Mon, 6 Jan 1997 04:07:57 -0600
To: dsp@tapr.org
Cc: "HF SIG  list   mailing"<hfsig@tapr.org>
Message-ID: <v03007815aef67c85613f@[192.168.1.2]>
Rendered body type: text/plain
```

The full URL for Johan's new upload is:
ftp://ftp.tapr.org/tapr/SIG/dsp/recent_uploads

Just to make the point again. If you upload something into any of the
UPLOAD areas on TAPR you must send e-mail to the upload area manager for
them to either move the file or make it readable.

This was necessary because TAPR had become the target of software pirates
that were using the TAPR system to transfer several hundred megs of files
on a regular basis in hidden dirs in the various upload areas. There are
programs on the Internet that look for sites that have T1 or better access
and open anonymous upload. Since TAPR.ORG sits on 3 T1s and some faster
networks, we became a good target for illegal usage. Since the change we
have had no problems from these pirate software people. It makes it a
little slower to get files available for everyone, but is necessary because
of others on the Internet and to ensure TAPR.ORG is not being used for
illegal activities.

Also, I just talked to Johan and we plan on moving the HF SIG upload area
over to the DSP area, because it makes sense. Eventually the DSP-93 and
other DSP projects will be under the DSP area. We will have to find a
full-time library manager to keep up with the area and also make sure that
major updates end up being reflected in the web pages. Tim is doing the
ftp area mgmt right now as a favor. If anyone is interested, please
contact me.

I am attaching the README file which is in the UPLOAD area. Please read it.

Cheers - Greg

TAPR.ORG is now setup as follows for all open anonymous upload areas, which now
includes the software_lib/UPLOAD area:

* No subdirs in the upload area can be created
* File names can only use the characters "A-Z","a-z","0-9",".","-", and "_".
* File names may not start with "." or "-"
* FTP users cannot delete, overwrite, or modify files once uploaded
* Files uploaded there are owned by the software_lib account with mode 0660.

This last statement means that anyone can still upload a file to an upload
area, but no one can read the uploaded files until someone (either aa5df
or any member of the ftp group) changes the permissions on the file
(and hopefully moves it somewhere else).

Tim Bagget, AA5DF, is now acting DSP SIG ftp manager. (aa5df@tapr.org)

When you upload a file to the software_lib upload area, you will need to
__send a message__ to 'aa5df@tapr.org'. In this way, Tim will know when
files are there and can either move them or make them readable until he
can move them.

Be sure to include the .TXT file describing the file uploaded. This helps
TAPR and others later know that the files contains. If you have a specific
area you want a file moved to, then let Tim know.

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd
-----

9. Re: Stanag 4529

From: "Bergzen Hakan, KR" <hakan.bergzen@enator.se>
Date: Mon, 6 Jan 1997 18:15:00 +0100
To: HFSIG at TAPR <hfsig@tapr.org>
Message-ID: <32D6B1C2@noak>
Rendered body type: text/plain

>Charles Brain wrote:
>
> If anybody wants a copy of this STANAG (It is the narrow bandwidth version
> of 4285 I have found an electronic copy and can send it to them. It is a
70k
> zipped file that consists of a Word document and a powerpoint one.
>
> I don't think it is a good idea to post it directly to the sig and I can't
> remember where I found it so I can't point you there!!

Hello Charles,

You probably found it at Peter Reynolds' homepage. Type this to download the
file:

http://ourworld.compuserve.com/homepages/pdreynolds/1240hz.zip

He has some other draft papers dealing with maritime LAN/HF gateways. (They
are quite big to print, though)

73/Hakan

10. Re: [HFSIG:1898] Re: 10m FM packet

From: Erik Skovgaard <Erik_Skovgaard@bc.sympatico.ca>
Date: Mon, 06 Jan 1997 22:26:59 -0800
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970107062659.006a6328@mail.bctel.ca>
Rendered body type: text/plain

At 03:33 05/01/97 -0600, you wrote:
>At 23:55 04/01/97 -0600, you wrote:
>>Sounds like a specific US issue. In Canada we are allowed to operate any
>>mode on 10 meters as long as we stay within a 20 kHz limit.
>>
>>OTOH - why waste a good band....
>>
>>73 de VE7MDL   ....Erik.
>>
>>
[stuff deleted]
>I am afraid I didn't look at the title of the message before I wrote HI.
>I was referring to the lower bands. On 10m I don't think it is a problem,
>but I am not familiar with the U.S regs. However comments have been made
>in the past on the Sig about the BW limit on the lower bands.
>
>By the way was there an HF blackout yesterday (SAT) or has my transceiver
>gone wrong ?

My log says that I had three contacts on RS-12 (15m up and 10 down), so
can't be all bad.
>
>- Charles
>
>
>
73 de VE7MDL   ....Erik.

11. Hello ..

From: "Frank Knoll" <fknoll@ibm.net>
Date: Wed, 08 Jan 97 21:09:47 +0100
To: "HFSIG" <hfsig@tapr.org>
Message-ID: <199701082015.UAA15891@pop01.ny.us.ibm.net>
Rendered body type: text/plain

i am working on OFDM (16 tones, 62.5hz freq. spacing DPSK or DQPSK)
with viterbi coding. The hardware platform is an 32c026 DSP and
an 68HC11 as the controller (equipment name = DSPCOM).
We (Walter Muth and i) will add as soon as possible a 32c542
to get more computational power for more complex coding.

The viterbi coder works (1/2 rate) and the synchronisation with a

special preamble for synchronisation is also working but not yet tested on shortwave.
The Synchronisation is able to capture a frequency differenc of -+250hz
which will be enough for the most amateur KW Transceivers. It
needs approx. 0.5sek for capture.

Now i have to define a simple protocoll, a more complex later.
First i will define a FEC protocoll due to its simplicity. Later
i will define a ARQ protocoll.

thats for now all folks

12. echo dsp sound card

From: Stephen PEARCE <spearce@ccu1.auckland.ac.nz>
Date: Thu, 9 Jan 1997 19:20:54 +1300 (NDT)
To: hfsig@tapr.org
Message-ID: <199701090620.TAA29696@ccu1.auckland.ac.nz>
Rendered body type: text/plain

Does anyone have a good description of the interface protocol
for the Echo DSP PSA sound card?
        I am trying to compile the cw250 and hfmodem code written
by KC7WW for psa sound cards .. but using turbo C
version 2 (because I want to run in a dos rather than a
windows os .. and the bcc version I have (3.1) is full
of bugs and seems to abort most compiles with run time errors i.e the
compiler binary is wrong).
I have been unable to find the source for the obj. files
that need to be linked with this code (psa.obj and psa1.obj)
and have had difficulty linking with my compiled code.

        In addition I want to use the echo card .. at address 0x0230
.. and it seems a little different from the orchid and cardinal cards.
All I have managed to do with it so far is read its id (E 3)
and reset it. I have not found any good description of it
via internet .. and ECHO speech coorporation does not
seem to be known there also.

        Any help gratefully accepted

        Thanks
        Stephen Pearce

13. Re: [HFSIG:1905] echo dsp sound card

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 09 Jan 1997 09:09:22 -0800
To: hfsig@tapr.org
Message-ID: <199701091657.IAA21088@PEAK.ORG>
Rendered body type: text/plain

Stephen,

Hi - Been a while since I heard from someone using one of those - they are
great performers and a pleasure to work with.

The software you are asking is somewhat different that what you are describing,
a) You have a later version of the ISA bus ASIC (the "ECHO" chip),
b) The ECHO card's arhictecture is designed to be Plug&Play cpmpliant where you
   need to be able to probe for the presence of the card, find the port address
   space and arm the right IRQ.

Look for PSATOOL2.ZIP at ftp://ftp.tapr.org/tapr/SIG/hfsig/upload/PSATOOL2.ZIP

The utilities are folded into one now and called "psautil" - C-sources are
included.
This utility will do all the above probing and will work with either the old
style ASIC (Cardinal/Orchid) or the newer ones (ECHO).

I would recommend you also pay a visit to Echo Speech on the web
(www.echospeech.com) and look at the software that they offer for free -
their example code will show you how to do it. Otherwise, Analog Devices has
the ASIC (older style) specs and source code for the card's DOS device
driver on their web site (www.analog.com - look under the 21xx family DSP).

Good luck with your project.

--Johan Forrer, KC7WW

>
>   Does anyone have a good description of the interface protocol
>for the Echo DSP PSA sound card?
>   I am trying to compile the cw250 and hfmodem code written
>by KC7WW for psa sound cards .. but using turbo C
>version 2 (because I want to run in a dos rather than a
>windows os .. and the bcc version I have (3.1) is full
>of bugs and seems to abort most compiles with run time errors i.e the
>compiler binary is wrong).
>I have been unable to find the source for the obj. files
>that need to be linked with this code (psa.obj and psa1.obj)
>and have had difficulty linking with my compiled code.
>
>   In addition I want to use the echo card .. at address 0x0230
>. and it seems a little different from the orchid and cardinal cards.
>All I have managed to do with it so far is read its id (E 3)
>and reset it. I have not found any good description of it
>via internet .. and ECHO speech coorporation does not
>seem to be known there also.
>
>   Any help gratefully accepted
>
>   Thanks
>   Stephen Pearce
>
>

14. Re: [HFSIG:1905] echo dsp sound card

From: Phil Karn <karn@laptop.ka9q.ampr.org>
Date: Thu, 9 Jan 1997 13:11:17 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <m0viRkr-000MFlC@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>by KC7WW for psa sound cards .. but using turbo C
>version 2 (because I want to run in a dos rather than a

You might consider DJGPP. This is a port of GCC (the GNU C compiler)
to a DOS environment. It does require a 386 or better as the compiler
is 32-bit only. It comes with a DOS extender so your program can use
all your memory under DOS.

The GNU assembler language is a different syntax than Borland, but
I have discovered that the support functions in the DJGPP library
are powerful and complete enough that I was able to simply discard much
of the asm code I had originally written for NOS.

Best of all, DJGPP is *free*.

Phil

15. Re: [HFSIG:1894] pactor

From: George Cserenyi <zsolt@direct.ca>
Date: Fri, 10 Jan 1997 12:33:48 -0800 (PST)
To: Alex Evonosky <alexevon@abraxis.com>
Cc: hfsig@tapr.org
Message-ID: <Pine.LNX.3.95.970110123306.12248H-100000@boxer.direct.ca>
Rendered body type: text/plain

Hi,
        How do you like it and how much was it?
73, George

On Wed, 1 Jan 1997, Alex Evonosky wrote:

> I recently purchased the PacCom PTC-II. I am wondering where do most of the
> Pactor sig run around? Wanted to test this unit out and see how it copies
> compared to the AEA-DSP232.
> Thank You in Advance.
>
> Best 73,
> de KB4LBX/Alex
> ---------
> Alex F. Evonosky, KB4LBX   OSCAR P3D U/V,V/U,L/U
> Alpharetta, GA / EM74   Packet LAN 145.07MHz
> WAS WAC ARRL AMSAT   ATV 1.2GHz/421.25MHz
> kb4lbx@amsat.org   160m-1.2GHz SSB/CW/Digital/SSTV
>   GRAPES 56kB Packet WAN
>   Active HF/VHF contester

```
>                            DX4WIN Computer Logging System
>                            W95SSTV SSTV software
>
>  ______ __________ _____ ________ ________ ____ __
>  ___  //_/___  __ )__  // /___  / ___  __ )__  |/ /
>  __  ,<   __  __  |_  // /___  /  __  __  |__    /
>  _  /| |  _  /_/ / /__  __/_  /____  /_/ / _     |
>  /_/ |_|  /_____/    /_/   /_____//_____/  /_/|_|
>
>
>
>
```

16. Better (then) Pentium?

From: George Cserenyi <zsolt@direct.ca>
Date: Fri, 10 Jan 1997 13:59:15 -0800 (PST)
To: HF SIG <hfsig@tapr.org>
Message-ID: <Pine.LNX.3.95.970110135335.13656B-100000@boxer.direct.ca>
Rendered body type: text/plain

Hi,
I know there are people on this list who are developing modems and decoders using the PC's own power, rather then using plug in dongles. This might be some interest to those too, who are about to upgrade their system.
Please read on,
George

ISRAELI-DEVELOPED INTEL PENTIUM PROCESSOR TO HANDLE MULTI-MEDIA

Intel Corp. has produced a new Pentium processor which was developed entirely in Israel, IDF Radio, GALEI TZAHAL reported. The MMX Pentium transfer integrates the capacity to transmit data from a Pentium computer with advanced multi-media capability, greatly improving computer performance.
The idea is the work of Intel's development center in Haifa, which began with a $100 million investment. Dov Fruman, chairman of Intel's operation in Israel, said he was proud of the local development, which has received wide international acclaim. Fruman added that computers will run faster and
will have an improved ability to handle complicated programs, as well as audio and video data.
According to GALEI TZAHAL, hundreds of millions of dollars worth of the product will be sold and it should attract additional investment in Israel.

17. ALE/High Speed HF Modem Software

From: "James M." <james119@iu.net>
Date: Sun, 12 Jan 1997 12:57:19 -0800
To: hfsig@tapr.org
Message-ID: <32D9502F.293@iu.net>
Rendered body type: text/plain

Question:

Does anyone have C-code to implement DSP to demodulate/decode 8-ary FSK ALE signals? Or 39-Tone or Single Tone HF Modem Signals (per MIL-STD-188). Code for any or all of the functions would help.

Thanks,

Jim

18. PSK modem vs PC + sound card or DSP engine?

From: "Peter Markham" <pmarkham@up.net>
Date: Tue, 14 Jan 1997 14:22:41 -0500
To: amsat-bb@amsat.org
Cc: hfsig@tapr.org
Message-ID: <9701141921.AA09831@btc1>
Rendered body type: text/plain

I am considering my options for satellite PSK operation in relationship to my interest in other digital modes, both hf and above. Considering the impending hardware improvements for mid priced PCs, including peripherals such as more versatile sound cards with greater processing power, is there an impending revolution that will make dedicated hardware modems obsolete, shortly, and software

modems a cost effective altenative to dedicated hardware?

 Brian Beezley and Phil Karn are two software wizards I am aware of
that are actively developing software modems for PCs. Guessing what
is possible, I am loathe to spend my hard earmed shekels on obsolete
hardware. Is there any objective insight among this fraternity that
may help remove much of the anxiety from my choices? Are software
modems for the masses, just around the corner? If so, what would the
computer hardware requirements be, looking into a crystal ball at
what the computer market is likely to offer me, next Winter?

                                        Pete/wa4hei

  The meek will not inherit the Earth.

19. Re: PSK modem vs PC + sound card or DSP engine?

From: Kory Hamzeh <kory@avatar.com>
Date: Tue, 14 Jan 1997 14:01:53 -0800 (PST)
To: Peter Markham <pmarkham@up.net>
Cc: amsat-bb@amsat.org, hfsig@tapr.org
Message-ID: <Pine.BSI.3.91.970114135612.22086A-100000@avatar.avatar.com>
Rendered body type: text/plain

Hi Peter,

I'm in the processes of trying to put such a "consortium" together to
develope a PC based TNC to support and HF and Sat type protocols. I have
started some discussion on the Usenet rec.radio.amateur.digital.misc
group with very little intersted from other. I was about to contact Phil
(I don't know Brian) and see if he would be intersted in such a thing. I
am willing to host the mailing list to kick off such discussion.

There has been some very nice work done by Johan Forrer (KC7WW) using PSA
DSP based sound cards. He has written AMTOR and PACTOR code. There a lot
of RTTY out there. I've seen some PBSK also. I am working on weak signal
CW recovery right now. I'm going through a huge learning curve trying to
understand DSP techniques.

Kory
AC6RN

On Tue, 14 Jan 1997, Peter Markham wrote:

>
>   I am considering my options for satellite PSK operation in
> relationship to my interest in other digital modes, both hf and
> above. Considering the impending hardware improvements for mid priced
> PCs, including peripherals such as more versatile sound cards with
> greater processing power, is there an impending revolution that will
> make dedicated hardware modems obsolete, shortly, and software
> modems a cost effective altenative to dedicated hardware?
>
> Brian Beezley and Phil Karn are two software wizards I am aware of
> that are actively developing software modems for PCs. Guessing what
> is possible, I am loathe to spend my hard earmed shekels on obsolete
> hardware. Is there any objective insight among this fraternity that
> may help remove much of the anxiety from my choices? Are software
> modems for the masses, just around the corner? If so, what would the
> computer hardware requirements be, looking into a crystal ball at
> what the computer market is likely to offer me, next Winter?
>
>   Pete/wa4hei
>
>
>   The meek will not inherit the Earth.
>

20. Re: [HFSIG:1912] Re: PSK modem vs PC + sound card or DSP engine?

From: Bill Carver <bcarver@magiclink.com>
Date: Tue, 14 Jan 1997 22:54:59 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199701150454.WAA05628@tapr.org>
Rendered body type: text/plain

Focus is Everything, and a TAPR commitment to hardware/software modem
platform for varioius HF modes is all it would take for me (and many other
silent folks, I suspect) to jump on that bandwagon so we aren't slaves,

locked into the latest "designer" digital mode.

Bill - W7AAZ

21. 39 tone Implementational questions

From: Charles Brain <chbrain@dircon.co.uk>
Date: Wed, 15 Jan 1997 10:10:22 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970115101022.00669bdc@popmail.dircon.co.uk>
Rendered body type: text/plain

Hello,

I need a bit of advice!

I am trying to implement a parallel tone modem. I have taken the 39 tone modem described in MIL-STD 188-110A Appendix B as a starting point. So far I have the transmit side working (as far as I can tell i.e it generates the right tones and the phase reversals during the preamble sequence appear correct) and have started on the receive side.

Does anybody have an idea on the best way to use the preamble tones to gain synchronisation. Currently I am DBPSK demodulating the 1800 Hz tone in the Phase 2 preamble to achieve frame sync. However I am throwing away the information contained in the other two tones (1125Hz and 2475Hz). What I am looking for is a computationally 'cheap' way to use all of this information.

Is it necessary to doppler track througout the entire frame or can I do doppler correction in the preamble phase and then apply this correction during the entire frame.

When extracting the phase change information what is the best way to do it (during data transmission)?
I am currently using an FFT then using @ = atan(Im/Re) to do this.
The atan approximation takes 58 cycles. Is there a better way ?

For anyone that doesn't have the MIL SPEC the modem uses a baud rate of 44.44 baud (22.5ms).
The tones are spaced 56.25 Hz apart. The preamble consists of three phases

Phase 1 = 4 unmodulated tones of 787.5Hz, 1462.5Hz, 2137.5Hz and 2812.5Hz.
Phase 2 = 3 phase modulated tones (phase advanced by 180deg on the boundry
of each signal element) 1125Hz, 1800Hz and 2475Hz.
Phase 3 = all 39 data tones and the doppler tone to act as a phase reference.

Thanks

Charles G4GUO

22. Re: PSK modem vs PC + sound card or DSP engine?

From: Douglas Braun <dougbert@hol.gr>
Date: Wed, 15 Jan 1997 17:06:15 -0200 (GMT)
To: amsat-bb@amsat.org, pmarkham@up.net
Cc: hfsig@tapr.org
Message-ID: <199701151906.RAA09678@prometheus.hol.gr>
Rendered body type: text/plain

Well, if you end up buying a dedicated modem, I
recommend the Motorola DSP56002 EVM card.
For $150 or so, this will do both 1200 BPS PSK and 9600 BPS
G3RUH, as well as regular 1200 BPS packet, and anything
that somebody writes the code for...

Douyg Braun
SV/N1OWU

23. Re: [HFSIG:1914] 39 tone Implementational questions

From: Rob Frohne <frohro@wwc.edu>
Date: Thu, 16 Jan 1997 11:58:04 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199701161958.LAA01541@adams.wwc.edu>
Rendered body type: text/plain

Hi Charles,

I don't have any answers to your question, but it sounds like you are doing great work. I'm wondering if you can let me know how you are doing the atan(y/x) operation. I have a need for that code that

should be coming up soon.

Thanks,

Rob KL7NA/W7
>
> Hello,
>
> I need a bit of advice!
>
> I am trying to implement a parallel tone modem. I have taken the 39 tone
> modem described in MIL-STD 188-110A Appendix B as a starting point. So far
> I have the transmit side working (as far as I can tell i.e it generates the
> right tones and the phase reversals during the preamble sequence appear
> correct) and have started on the receive side.
>
> Does anybody have an idea on the best way to use the preamble tones to gain
> synchronisation. Currently I am DBPSK demodulating the 1800 Hz tone in the
> Phase 2 preamble to achieve frame sync. However I am throwing away the
> information contained in the other two tones (1125Hz and 2475Hz). What I am
> looking for is a computationally 'cheap' way to use all of this information.
>
> Is it necessary to doppler track througout the entire frame or can I do
> doppler correction in the preamble phase and then apply this correction
> during the entire frame.
>
> When extracting the phase change information what is the best way to
> do it (during data transmission)?
> I am currently using an FFT then using @ = atan(Im/Re) to do this.
> The atan approximation takes 58 cycles. Is there a better way ?
>
> For anyone that doesn't have the MIL SPEC the modem uses a baud rate of
> 44.44 baud (22.5ms).
> The tones are spaced 56.25 Hz apart. The preamble consists of three phases
>
> Phase 1 = 4 unmodulated tones of 787.5Hz, 1462.5Hz, 2137.5Hz and 2812.5Hz.
> Phase 2 = 3 phase modulated tones (phase advanced by 180deg on the boundry
>   of each signal element) 1125Hz, 1800Hz and 2475Hz.
> Phase 3 = all 39 data tones and the doppler tone to act as a phase reference.
>
>
> Thanks
>
> Charles G4GUO
>
>

24. Re: [HFSIG:1916] Re: 39 tone Implementational questions

From: Charles Brain <chbrain@dircon.co.uk>
Date: Thu, 16 Jan 1997 22:34:53 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970116223453.00678cb0@popmail.dircon.co.uk>
Rendered body type: text/plain

At 14:05 16/01/97 -0600, you wrote:
>Hi Charles,
>
>I don't have any answers to your question, but it sounds like you are
>doing great work. I'm wondering if you can let me know how you are
>doing the atan(y/x) operation. I have a need for that code that
>should be coming up soon.
>
>Thanks,
>
>Rob KL7NA/W7
>>
>
Well I am using a routine from analogue devices, I have attached the code.
I have produced a lookup table approach that uses the technique described
in Ferking (a good book) page 221.

Regards Charles

25. RE: Tiny two MK - 2 Capacitor C 23 Question

From: Gwyn Reedy <gwyn@paccomm.com>
Date: Fri, 17 Jan 1997 13:54:13 -0500
To: "'Brian Boccardi'" <n2mpm@worldnet.att.net>
Cc: "'HFSIG'" <hfsig@tapr.org>, "'APRSSIG'" <aprssig@tapr.ofrg>,

"'NETSIG'"
 <netsig@tapr.org>, "'BBSSIG'" <bbssig@tapr.org>,
"'Bitbucket'"
 <bitbucket@primenet.com>,
"'Ham-Tech'"
 <owner-ham-tech@majordomo.netcom.com>
Message-ID: <01BC047D.ECF47C80@gwyn>
Rendered body type: text/plain

The report you received is correct. Let me tell you more about which models are affected and what should be done about it.

Amateur radio models affected are the TINY-2 MK-2, TNC-NB96, and IPR-NB96 which all incorporate the TINY-2 circuit board.

The TINY-2 pre-MK-2 version is not affected. Those models use the MAX-231 RS-232 chip. The mismarked capacitor crept into the design when the MK-2 changes were made, which included adding the capability to use the MAX-232 RS-232 chip. (The circuit board accepts either the 232 or 231, but all have been built with the 232).

Another way to identify the boards with the error is to examine the circuit board revision number. Revisions 1.7, 1.8, and 1.81 contain the error. Revisions 1.6 and prior use the MAX-231, and Revisions 1.9, 1.91 and 1.92 have corrected the problem. Some of the final production run of 1.81 boards may have the capacitor correctly installed (the reverse of the board markings) as the problem was identified late in the 1.81 production run as work was done on rev. 1.9. Visual inspection will confirm if this is the case.

This capacitor problem is of the nature of 'you ought to fix it the next time it is in for a major tune-up' and not of the 'park it immediately and tow it to the garage' type. In production we usually install capacitors with a significantly higher working voltage than what is applied to them and they have generally worked just fine in these mismarked boards, even though reversed in polarity. Nothing is going to be damaged if the capacitor is left in backwards, except maybe the capacitor itself. There are no reports of any capacitor explosions or other bad effects.

Repair instructions: Measure the voltage on pin 6 of U2 (MAX232). It should be in the range of 8-10 VDC. Remove C23 and reinstall with the negative lead in the + marked hole. Measure the voltage on pin 6 again. If still within the 8-10 VDC range then C23 was not damaged.

If the pin 6 voltage is out of range either before or after reversing C23, the capacitor should be replaced.

Gwyn Reedy, W1BEL
PacComm Packet Radio Systems

gwyn@paccomm.com
http://www.paccomm.com

From: Brian Boccardi[SMTP:n2mpm@worldnet.att.net]
Sent: Thursday, January 16, 1997 1:17 PM
To: techspt@paccomm.com
Subject: Tiny two MK - 2   Capacitor C 23 Question

Hello ,
I am a proud owner of a Tiny-2 Mk2, recently i received information about a possible problem with C23. I have serial no 20427 board rev is 1.81. I was told that c 23 may be in backwards and the polarity mark on the board is wrong. The tnc is not in use right now but would like to get it back on the air. (A older tiny 2 is standing in for it)

N2MPM
Brian Boccardi
Avid MacAPRS User

26. GR impedance bridge

From: Larry Bush <lbush@eramp.net>
Date: Sat, 18 Jan 1997 05:51:19 -0600
To: hfsig@tapr.org
Message-ID: <2.2.32.19970118115119.0069bae0@eramp.net>
Rendered body type: text/plain

I'm looking for a used General Radio model 916A RF impedance bridge.

Larry Bush, W5NCD
359 Arrowhead Point
Waco, Texas 76712
phone 817-848-5155
Fax # 817-299-1040

27. Re: [HFSIG:1917] Re: 39 tone Implementational questions

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sun, 19 Jan 1997 06:35:46 -0500
To: hfsig@tapr.org
Message-ID: <32E20712.7E2F@worldnet.att.net>
Rendered body type: text/plain

Charles Brain wrote:
>
> At 14:05 16/01/97 -0600, you wrote:
> >Hi Charles,
> >
> >I don't have any answers to your question, but it sounds like you are
> >doing great work. I'm wondering if you can let me know how you are
> >doing the atan(y/x) operation. I have a need for that code that
> >should be coming up soon.
> >
> >Thanks,
> >
> >Rob KL7NA/W7
> >>
> >
>   DIVS AY1,AX0;
>   DIVQ AX0; DIVQ AX0; DIVQ AX0;
>   DIVQ AX0; DIVQ AX0; DIVQ AX0;
>   DIVQ AX0; DIVQ AX0; DIVQ AX0;
>   DIVQ AX0; DIVQ AX0; DIVQ AX0;
>   DIVQ AX0; DIVQ AX0; DIVQ AX0;

YEEOOOWWWW! That will eat you alive. Consider doing a table lookup
where
you trade memory for a lot faster speed. Table lookup approaches for
atan are more than usually adequate and there are formulae for taking a
table lookup output that "gets you close" and then uses an iterative
approach to remove some or most of the residual error. Maybe someone
has
a table lookup algorithm they would care to upload.

--
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

28. Preamble Detection

From: "Frank Knoll" <fknoll@ibm.net>
Date: Sun, 19 Jan 97 22:58:17 +0100
To: "HFSIG" <hfsig@tapr.org>
Message-ID: <199701192159.VAA99211@pop01.ny.us.ibm.net>
Rendered body type: text/plain

Hello,

is there anybody with experience about a robust
preamble detection algorithm for OFDM.

I use a simple autocorrelation function to detect
my OFDM preamble which is used to get
the timing synchronisation. (all 16 subcarriers modulated
with a alternate Symbols). This method works good,
but it has a problem if there is one or more
jamming carriers (due to the us of autocorrelation in the RX).

I want to try out the use of Cross-Correlation in the RX,
but may be, there is anybody with a better idea.

Thanks
best regards
Frank Knoll

==
* Frank Knoll
* Spessartstr. 12
* D-63571 Gelnhausen-Meerholz
* Germany
* Tel.: +49 6051 969116
* E-Mail: fknoll@ibm.net   or dl2fch@amsat.org
==

29. Re: [HFSIG:1921] Preamble Detection

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 21 Jan 1997 18:28:29 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970121182829.00665c64@popmail.dircon.co.uk>
Rendered body type: text/plain

At 16:05 19/01/97 -0600, you wrote:
>Hello,
>
>is there anybody with experience about a robust
>preamble detection algorithm for OFDM.
>
>>
Hello Frank,

I hope you have better luck than I did! I asked a similar question a week
back and got 2 replies, no answers just questions !

Still it wouldn't be fun if we didn't have to sweat a bit.

I am trying to develop a 39 tone modem along the lines of the MIL STD one,
similar to what you are doing. I have the transmit side going and am
experimenting with the best way to recover the preamble sequence. Hopefully
I can sort something out by the end of the week.

I would like to hear how you are getting on when you have the time.

73 Charles G4GUO

30. Re: [HFSIG:1922] Re: Preamble Detection

From: "Frank Knoll" <fknoll@ibm.net>
Date: Tue, 21 Jan 97 22:10:12 +0100
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <199701212114.VAA84041@pop01.ny.us.ibm.net>
Rendered body type: text/plain

On Tue, 21 Jan 1997 12:29:48 -0600 (CST), Charles Brain wrote:

>Hello Frank,
>
>I hope you have better luck than I did! I asked a similar question a week
>back and got 2 replies, no answers just questions !
>
>Still it wouldn't be fun if we didn't have to sweat a bit.
>
>I am trying to develop a 39 tone modem along the lines of the MIL STD one,
>similar to what you are doing. I have the transmit side going and am
>experimenting with the best way to recover the preamble sequence. Hopefully
>I can sort something out by the end of the week.
>
>I would like to hear how you are getting on when you have the time.
>
>73 Charles G4GUO
>
>

Hello Charles,

i am working on an OFDM Modem with 16 tones, 50 baud, 62.5 hz
subcarrier spaceing and 2DPSK or QDPSK on each carrier.
I also tested viterbi coding (rate 1/2). I do not use a Doppler
Carrier or similar things.

For Peamble Detection i used a simple autocorrelation
which detects the part one of the preamble to synchronisize the timing.
The second part of the preamble is used to calculate and correct the
frequency error bettween RX and TX. It can correct a freq. error
up to +-250 hz. The whole preamble is approx. 0.5sek long.

I tested the modem some days ago with weak signals but without
viterbi coding. The TX transmitted simple packets, each with
a preamble for timing and freq. synchronisation and a text
like "the quick brown ...", so it was no problem to detected
any error on the RX-Display.

 I t works, but not very good, but it works. I would say for aprox
50% of the packets the RX was able to demodulate the
Data (the Text), but with some erros. Well, the next step
would be to ernable the coding.

To be able to compare different algorithm for
demodulation or preamble detection, i need a channel simulator
to get a known channel every time i need it. Thats
the reason why i am now working on a channel Simulator
with Alex (DL8AAU) for DSPCOM. (DSPCOM is our hardware
 we us. TMS320C26 DSP and 68HC11 as Controller)

At least, I hope there is anybody who knows something
about preablme detection.

Thats what i did, mybe we can help each other.

best regards
Frank Knoll

===
* Frank Knoll
* Spessartstr. 12
* D-63571 Gelnhausen-Meerholz
* Germany
* Tel.: +49 6051 969116
* E-Mail: fknoll@ibm.net   or dl2fch@amsat.org
===

31. AX25 PID registry?

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Thu, 23 Jan 1997 11:17:18 +0100
To: hfsig@tapr.org
Message-ID: <32E73AAE.3B61@ife.ee.ethz.ch>
Rendered body type: text/plain

offtopic... but...

does anyone know who is responsible for coordinating AX.25
PID's? I wasn't successful so far finding someone responsible for
this... anyone from TAPR?

We need two PIDs for the implementation of a VJ like IP
header compression...

73s
Tom

32. Packet and Soundblaster any progs?

From: vk7zo@calvados.apana.org.au (Graham Ranft)
Date: Thu, 23 Jan 97 07:32:39 +1100
To: hfsig@tapr.org
Message-ID: <5oZB4D1w165w@calvados.apana.org.au>
Rendered body type: text/plain

Hi, does anyone have, or know of, a packet program that uses the
soundblaster instead of a dedicated modem for use on an ibm dos/windows machine?

Mant tnx! & 73!
Please reply to vk7zo@calvados.apana.org.au

--
vk7zo@calvados.apana.org.au (Graham Ranft)
APANA, Hobart, Tasmania: 03 6234 8530

33. Re: [HFSIG:1924] AX25 PID registry?

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Thu, 23 Jan 1997 09:29:05 -0600
To: hfsig@tapr.org
Cc: "Paul Newland"<paul.newland@lucent.com>
Message-ID: <v03007800af0d332310ff@[208.134.134.40]>
Rendered body type: text/plain

The ARRL Future Systems Committee has responsibility now that the ARRL
Digital Committee was closed down last year. TAPR is helping to keep what
the list is, which can be viewed on the AX.25 page
http://www.tapr.org/tapr/html/ax25.html

You will need to contact Paul Rinaldo <prinaldo@interramp.com> on the subject.

Cheers - Greg, WD5IVD

>offtopic... but...
>
>does anyone know who is responsible for coordinating AX.25
>PID's? I wasn't successful so far finding someone responsible for
>this... anyone from TAPR?
>
>We need two PIDs for the implementation of a VJ like IP
>header compression...
>
>73s
>Tom

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd
-----

34. Re: [HFSIG:1925] Packet and Soundblaster any progs?

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Thu, 23 Jan 1997 16:47:17 +0100
To: hfsig@tapr.org
Message-ID: <32E78805.7042123E@zeus.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

Graham Ranft wrote:
>
> Hi, does anyone have, or know of, a packet program that uses the
> soundblaster instead of a dedicated modem for use on an ibm dos/windows machine?

Yes, sure. See: http://www.ife.ee.ethz.ch/~sailer/pcf/

73' Alexander

--
------------------------------+---------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| u. Elektroakustik | Fax : +49-6151-165545
|
| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de
|
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
------------------------------+---------------------------------------

35. 56002evm memory ?

From: zs6awk@global.co.za (Danie Brynard)
Date: Sun, 26 Jan 1997 21:36:28 +0200
To: hfsig@tapr.org
Message-ID: <199701262334.VAA11510@lin01.global.co.za>
Rendered body type: text/plain

TO DSP56002EVM developers:

When the evm is in 16k SRAM mode: is the ram from 16k to 32k mirrored from

0..16k ? I see flash.asm uses that area (FC00) but it works in 16k mode.

Danie zs6awk

++++++++++++++++++++++++++++++++
Danie Brynard ZS6AWK
Email: zs6awk@global.co.za
PO Box 15133
Centurion
0140
South Africa
GRID LOC: KG44CE
Tel/Fax: (2712) 6640330
QRV: 160m..23cm
++++++++++++++++++++++++++++++++

36. Pactor-II tests

From: Gwyn Reedy <gwyn@paccomm.com>
Date: Thu, 30 Jan 1997 16:59:13 -0500
To: "'HFSIG'" <hfsig@tapr.org>
Cc: "'Russ Chasteen'" <RChasteen@maf.org>,
"'Jerome Lacap'"
 <jerome@dove.com.ph>,
"'Dave Pedersen'" <DPedersen@maf.org>,
"'Tom Rink'" <Rink@em.uni-frankfurt.de>
Message-ID: <01BC0ECE.EC93EAE0@gwyn>
Rendered body type: text/plain

Bulletin to radio amateurs and short wave listeners with Pactor or
Pactor-II capability

Marine Radio Network (marine stations WLO, KLB, WSC) will be transmitting
Pactor test signals this weekend from WLO in Mobile Alabama. The
frequencies in use will be:
4256.0
6416.0
8473.5
13051.5
16997.6

I believe they will be transmitting Pactor-I FEC protocol this weekend with
other tests on subsequent weekends.

The text being sent has no meaning - disregard content.

Gwyn Reedy, W1BEL
PacComm Packet Radio Systems

gwyn@paccomm.com
http://www.paccomm.com

37. (no subject)

From: Dave Maples <Dave.Maples@ssc.nasa.gov>
Date: Fri, 31 Jan 1997 08:18:04 -0600
To: hfsig@tapr.org
Message-ID: <s2f1ac26.071@wpogate.ssc.nasa.gov>
Rendered body type: text/plain

How do I subscribe to this list?

38. Is the KAM upgradable?

From: N0AOT@aol.com
Date: Fri, 31 Jan 1997 13:39:46 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <970131133942_1893963286@emout10.mail.aol.com>
Rendered body type: text/plain

Hi All,

Does anyone know if the original KAM All Mode TNC can be upgraded to the
current KAM Plus with firmware revision 8? Are the motherboards the same,
differing only in the firmware?

Thanks much,

--Bob Carlson, n0aot

39. Re: [HFSIG:1931] Is the KAM upgradable?

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Fri, 31 Jan 1997 16:03:12 -0500
To: hfsig@tapr.org
Message-ID: <199701312103.QAA00198@aus-d.mp.campus.mci.net>
Rendered body type: text/plain

At 12:45 PM 1/31/97 -0600, you wrote:
>Hi All,
>
>Does anyone know if the original KAM All Mode TNC can be upgraded to the
>current KAM Plus with firmware revision 8? Are the motherboards the same,
>differing only in the firmware?
>
>Thanks much,
>
>--Bob Carlson, n0aot
>
>

Bob,

I have an early version KAM and it currently is at 8.0. There is an upgrade
that took place about v 6 which required an add on card. It added battery
backed RAM. I believe the KAM Plus included this mod on the main board,
i.e. the boards are slightly different, but as I understand it the PROMs are
the same.

Don

40. Re: [HFSIG:1932] Re: Is the KAM upgradable?

From: George Cserenyi <zsolt@direct.ca>
Date: Fri, 31 Jan 1997 14:55:22 -0800 (PST)
To: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Cc: hfsig@tapr.org
Message-ID: <Pine.LNX.3.95.970131145425.1766C-100000@cirmos.ci.ca>
Rendered body type: text/plain

Hi Don,
        have u had to send ur KAM in for the uograde?
How much did the upgrade cost?
George

On Fri, 31 Jan 1997, Donald E. Haselwood wrote:

> At 12:45 PM 1/31/97 -0600, you wrote:
> >Hi All,
> >
> >Does anyone know if the original KAM All Mode TNC can be upgraded to the
> >current KAM Plus with firmware revision 8? Are the motherboards the same,
> >differing only in the firmware?
> >
> >Thanks much,
> >
> >--Bob Carlson, n0aot
> >
> >
>
> Bob,
>
> I have an early version KAM and it currently is at 8.0. There is an upgrade
> that took place about v 6 which required an add on card. It added battery
> backed RAM. I believe the KAM Plus included this mod on the main board,
> i.e. the boards are slightly different, but as I understand it the PROMs are
> the same.
>
> Don
>
>
>

41. Re: [HFSIG:1931] Is the KAM upgradable?

From: "Roderick W. Hart" <rwhart@ix.netcom.com>
Date: Fri, 31 Jan 1997 19:22:09 -0500
To: hfsig@tapr.org
Message-ID: <32F28CB1.6CAB@ix.netcom.com>
Rendered body type: text/plain

N0AOT@aol.com wrote:
>
> Hi All,

>
> Does anyone know if the original KAM All Mode TNC can be upgraded to the
> current KAM Plus with firmware revision 8? Are the motherboards the same,
> differing only in the firmware?
>
> Thanks much,
>
> --Bob Carlson, n0aot

Roger. I have one of the original KAMs and it has been upgraded several
times. It is now at the latest revision (8.0).

February 1997

Original source file: hfsig.9702
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9702

1. Re: The ARRL Letter Vol 16, No 5

From: KB6ZVP@aol.com
Date: Sat, 1 Feb 1997 20:28:57 -0500 (EST)
To: elindquist@arrl.org, amsat-bb@amsat.org, netsig@tapr.org, tapr-bb@tapr.org, hfsig@tapr.org, bbssig@tapr.org, fbb@f6fbb.org, rick@janrix.com
Message-ID: <970201202856_1047856303@emout06.mail.aol.com>
Rendered body type: text/plain

No, it is not off topic. It affects us all. The future is now and if we don't take action, there won't be any Ham radio to talk about.

In a message dated 97-02-01 07:23:18 EST, the ARRL writes:

<< The Intermediate CW test would be 5 words per minute (the same as the current Tech Plus requirement), but the committee proposed that the General class CW requirement be set at 10 wpm. There still would be no additional CW

exam for the Advanced ticket, nor would there be any change in the 20-wpm requirement for the Extra. Exams for all classes would include a return to a

sending test and the requirement for one minute of solid copy during a five-minute test--instead of the current method that tests on the content of

the CW text. >>

Why? Why? Why?

Why does the League insist on maintaining this archaic link to the past? The only organization on the planet stuck in this rut is the League. Militaries, Governments, and other (mostly) advanced life forms have recognized the need to progress, yet the League continues to walk backwards.

Why not test on one's ability to operate voice? Why not test on one's ability to operate packet? How about ATV?

Why are you even considering testing on the ability to use one mode instead of all modes? Could your answer be "because we have always done it that way". Come on members of the League - join the 21st Century.

The code is dead, and Amateur radio will die if you/we do not advance the science and the membership. This antiquated tie to the code is forcing more and more of us away. I will not upgrade to General or higher because of the code requirement. I have no interest in it. My interest lies in more advanced methods of communication. Yet, I do not see a requirement for Hams to learn spread spectrum just to get a license. I have nothing against the code itself, just the requirement to test for it.

A sending test for code? Get real folks. Now you are really walking backwards in time. Go ahead, drive off the rest of the folks who had any interest in radio. When my son grows up, he will just throw my inherited $100,000 station in the junk pile. There is no test for the Internet and Ham radio cannot compete with this far superior mode of communication (AND it is in the classroom, the work place, every house in America will have it soon).

Come on tell me - what can I say to all the kids out there even remotely toying with the idea of getting into radio? They ask why? Why should I do that AND take a TEST, when I can do all of it on the Internet in real time for free and no test. What possible intelligence can there be behind an idea to step back 40 years in time and scare off all prospective new operators and stiffle this service. (Maybe the Directors have a hidden agenda all their own)

Does the membership know you are spending their money to study this project? I gave up my membership because of this stuck-in-the-mud-go-nowhere-philosophy that the Directors have. You haven't done anything progressive in years, and this is just another fine example of why the dinosaurs died.

Open your eyes and see the light - your killing us off! Or make the call to Congress and tell them to go ahead and sell off the rest of our frequencies, the League has spoken again.

By the way - this message went out to over 500,000 people in less than 15 seconds. No static, no drift, no fading, no fuss. And I know it got there accurately, without tuning, without antenna's, without TVI or RFI, and without the XYL and neighbors nagging. I should tell my son to learn the code because.....?

73 and standing by for the flames,

Mark, kb6zvp@aol.com

2. Re: The ARRL Letter Vol 16, No 5

From: "Larry W. Johnson" <larryj@cetlink.net>
Date: Tue, 14 Jan 1997 21:57:12 -0500
To: KB6ZVP@aol.com
Cc: elindquist@arrl.org, amsat-bb@amsat.org, netsig@tapr.org, tapr-bb@tapr.org, hfsig@tapr.org, bbssig@tapr.org, fbb@f6fbb.org, rick@janrix.com, onemm@cetlink.net
Message-ID: <32DC4788.2216@cetlink.net>
Rendered body type: text/plain

KB6ZVP@aol.com wrote:

>
>   No, it is not off topic. It affects us all. The future is now and if we
> don't take action, there won't be any Ham radio to talk about.
>
> In a message dated 97-02-01 07:23:18 EST, the ARRL writes:
>
> << The Intermediate CW test would be 5 words per minute (the same as the
> current Tech Plus requirement), but the committee proposed that the General
> class CW requirement be set at 10 wpm. There still would be no additional CW
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> exam for the Advanced ticket, nor would there be any change in the 20-wpm
> requirement for the Extra. Exams for all classes would include a return to a
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> sending test and the requirement for one minute of solid copy during a
> five-minute test--instead of the current method that tests on the content of
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> the CW text. >>
>
>   Why? Why? Why?
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>   Why does the League insist on maintaining this archaic link to the past?
> The only organization on the planet stuck in this rut is the League.
> Militaries, Governments, and other (mostly) advanced life forms have
> recognized the need to progress, yet the League continues to walk backwards.
>   Why not test on one's ability to operate voice? Why not test on one's
> ability to operate packet? How about ATV?
>   Why are you even considering testing on the ability to use one mode instead
> of all modes? Could your answer be "because we have always done it that
> way". Come on members of the League - join the 21st Century.
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> science and the membership. This antiquated tie to the code is forcing more
> and more of us away. I will not upgrade to General or higher because of the
> code requirement. I have no interest in it. My interest lies in more
> advanced methods of communication. Yet, I do not see a requirement for Hams
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> stiffle this service. (Maybe the Directors have a hidden agenda all their
> own)
>   Does the membership know you are spending their money to study this
> project? I gave up my membership because of this
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> done anything progressive in years, and this is just another fine example of
> why the dinosaurs died.
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> frequencies, the League has spoken again.
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> without the XYL and neighbors nagging. I should tell my son to learn the
> code because......?
>
> 73 and standing by for the flames,
>

> Mark, kb6zvp@aol.com

Hello Mark;

My name is Larry Johnson and my Call is W7LJ. I have been a Ham for 35 years and have been licenced in several countries,some with less stringent requirements and some with very difficult requirements.

First of all Ham Radio is a Hobby,it is something I enjoy,I derive pleasure from it.I was not forced into it and no one has ever pressured or coerced me into participating into any activity related to Amatuer Radio.

You as an American have the right to your opinion and the right to voice your opinion to anyone one that will listen to it.I assure you that from personal experiences in other countries that over half of the other people in the world do not share that privilege.

You state that backwards thinking on the part of the ARRL is killing Ham Radio.I do not agree with all of the positions that The ARRL takes but I remain a member because I feel they are the only Link between us and The FCC.

If you will research the facts for the last twenty years your will find the requirement to obtain a Ham Licence have been getting progressively Easier.Also since the the emplementation of The VE Program and the Codeless License the number of licensed hams has almost doubled.

As far as the new technology " G R E A T " ,I average about 200 hours a month on the internet. I still have time for Ham Radio,especially CW.

Just remember opinions are the most common thing around,everyone has one but every one forunately does not have to share the same one.

Larry W. Johnson (W7LJ)

3. Re: [HFSIG:1936] Re: The ARRL Letter Vol 16, No 5

From: "James M." <james119@iu.net>
Date: Sun, 02 Feb 1997 00:22:04 -0800
To: hfsig@tapr.org
Message-ID: <32F44EAC.7E95@iu.net>
Rendered body type: text/plain

Didn't anybody see the movie "Independence Day"? If it hadn't been for Morse code (after the aliens killed all our satellites) the planet would have been lost!

73, and long live Morse code, where good copy in negative SNRs can be achieved!

Jim N4BE

Larry W. Johnson wrote:
>
> KB6ZVP@aol.com wrote:
> >
> > No, it is not off topic. It affects us all. The future is now and if we
> > don't take action, there won't be any Ham radio to talk about.
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> > Does the membership know you are spending their money to study this
> > project? I gave up my membership because of this
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> > Mark, kb6zvp@aol.com
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> a month on the internet. I still have time for Ham Radio,especially CW.
>
> Just remember opinions are the most common thing around,everyone has
> one but every one forunately does not have to share the same one.
>
> Larry W. Johnson (W7LJ)

4. Re: [HFSIG:1935] Re: The ARRL Letter Vol 16, No 5

From: Bill Carver <bcarver@magiclink.com>
Date: Sat, 1 Feb 1997 23:39:33 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199702020539.XAA23175@tapr.org>
Rendered body type: text/plain

>don't take action, there won't be any Ham radio to talk about.

If ham radio is obsolete get a license for Internet and quit worrying about it. If you don't believe in 20 wpm, so be it...RTTY/AMTOR/Clover don't exist in the bottom 25 KHz anyway, just forget it, it isn't going to convey any useful additional spectrum to you anyway.

---

5. Intrigued by CW filter

From: zs6awk@global.co.za (Danie Brynard)
Date: Sun, 02 Feb 1997 07:32:58 +0200
To: hfsig@tapr.org
Message-ID: <199702020930.HAA25568@lin01.global.co.za>
Rendered body type: text/plain

Hi fellow dsp progammers...

I have been intrigued* by the DSP CW filters on the latest DSP HF radio's. These filters can be varied contineously from say 50Hz to 2000Hz. Can anybody give me a clue on how this is done ?

I have written some CW filters for HF for the DSP56002EVM (see cwawk.zip on upload) but was using a set of FIR filter coef's for a set of filters. Is there perhaps a DSP algorithm let allows one to make a virtually contineously variable FIR filter ? I would like to implemented this on my 56002 + hf radio and elevate it to 'dsp rig' status :-)) The 56002EVM has limited memory so one cannot load many many coef sets.

Yours sincerely

Danie zs6awk

* My english is not too good, I had to look up the word intrigued in a dictionary, it says it means appeal, arouse, seduce, lure, tantalise... :-)

Perhaps this is what new radio's do to us hams...

++++++++++++++++++++++++++++++++
Danie Brynard ZS6AWK
Email: zs6awk@global.co.za
PO Box 15133
Centurion
0140
South Africa
GRID LOC: KG44CE
Tel/Fax: (2712) 6640330
QRV: 160m..23cm
++++++++++++++++++++++++++++++++

---

6. Re: The ARRL Letter Vol 16, No 5

From: Thomas Ringate <ThomasLRingate@ij.net>
Date: Sun, 02 Feb 1997 16:17:17 -0500
To: larryj@cetlink.net
Cc: KB6ZVP@aol.com, elindquist@arrl.org, amsat-bb@amsat.org, netsig@tapr.org, tapr-bb@tapr.org, hfsig@tapr.org, bbssig@tapr.org, fbb@f6fbb.org, rick@janrix.com, onemm@cetlink.net
Message-ID: <32F5045C.148B@ij.net>
Rendered body type: text/plain

Larry W. Johnson wrote:

> Just remember opinions are the most common thing around,everyone has
> one but every one forunately does not have to share the same one.

My opinion is that as a responsible citizen of the USA, it's time to abolish amateur radio, and give the freq. back to the citizens of the USA. There are over 300 million citizens of the USA, but only a handful receive any use of this precious asset. How can any of us look at our neighbors and say we are making good use of an asset worth billions of $$ to the USA, and we abuse it like we do.

We should all be ashamed. We do not provide a service that is worth

the cost amateur radio is costing the rest of the non-amateur population
of this country. If the facts were presented in a national election,
and this were put to a vote of the general poplualtion, amateur radio
would be history never again to be heard from.

There is no justification to reserve a billion dollar asset for the
jokers who are using it.

Oh yes, I have also been an amateur since 1957, but I have been a
citizen of the USA since 1944 and somehow that responsibility weighs
more heavily than the simple enjoyment I get from talking on the radio
every now and then.

Keep CW, in fact make it 50 words per minute for the novice. Let's
get this over with as fast as we can so all 300+ million of us can get
some use out of this asset.

If we each had to pay a license fee to compensate for the lost
revenue this spectrum is worth, Bill Gates would be the only person
who could afford a license.

My vote is to end amateur radio and make a valid use of this
precious spectrum for the good of all, not just a few old fuddy duddies.

7. Re:The ARRL Letter Vol 16, No 5

From: Phillip Conza <phil@zl2tze.mlb.planet.gen.nz>
Date: Sun, 2 Feb 1997 21:14:22 +1200
To: hfsig@tapr.org
Message-ID: <32f5d482.zl2tze@zl2tze.mlb.planet.gen.nz>
Rendered body type: text/plain

HI Everyone,
             Another Country Lays Dinosaur To Rest..

Taken from the Packet Network in NZ submitted by Dave ZL2UTL..

The following is taken from the Sunday Star Times dated 2 Feb 1997

"France's coastguard has tapped out its final Morse Code messag,
dropping the 157 year old radio transmitted sounds for modern
satellite and radiotelephone transmissions. "Call all. This is our
last cry before our eternal silence" was the message from Le
Conquest, Britanny, to anybody still listening.

Over and Out!

73 Phillip ZL2TZE--
Phillip Conza
AX25   ZL2TZE @ ZL2TZE.#73.NZL.OC
E-Mail private   phil@zl2tze.mlb.planet.gen.nz

8. Re: [HFSIG:1935] Re: The ARRL Letter Vol 16, No 5

From: Chuck K3FT <k3ft@erols.com>
Date: Sun, 02 Feb 1997 17:13:08 -0500
To: hfsig@tapr.org
Message-ID: <32F51174.7804@erols.com>
Rendered body type: text/plain

KB6ZVP@aol.com wrote:

 it is not off topic. It affects us all. The future is now and if we
don't take action, there won't be any Ham radio to talk about.
>
> In a message dated 97-02-01 07:23:18 EST, the ARRL writes:
>
> << The Intermediate CW test would be 5 words per minute (the same as the (rest of text snipped under
Bandwidth Reduction Act of 1996)

de K3FT - That is the ARRL's position and not the U.S. Governement or
anyone elses' position.

 Why? Why?
>
>   Why does the League insist on maintaining this archaic link to the past? de k3ft - because the ITU
regs require a code proficiency to
operate below 30 Mhz.

> The only organization on the planet stuck in this rut is the League.(de k3ft NOT TRUE. Many
governments, private entities, and individuals

still use CW. Yes, it is old and outdated (in terms of throughput and
bulk-handling abilities) but still a viable communications medium.

>Militaries, Governments, and other (mostly) advanced life forms have
> recognized the need to progress, yet the League continues to walk backwards. (the League promotes 'new
 and advanced' comm systems - PACTOR,
AMTOR, APRS, etc etc.etc..)

The military DOES still teach CW for certain types of military ops.

>   Why not test on one's ability to operate voice? .. packet? .. ATV?de k3ft - how about operational
testing? proper ettiquette, procedure,
etc?) They do ask questions about all those modes and more.

>   Why are you even considering testing on the ability to use one mode instead of all modes? Could
your answer be "because we have always done
it that way". Come on members of the League - join the 21st Century.

de k3ft - testing required by ITU regs. they test on other modes.

>   The code is dead, (NOPE... OLD AND OUTDATED.. not the same thing) and Amateur radio will die if
you/we do not advance the
> science and the membership. (de k3ft - being done as we speak. by many groups - TAPR with APRS for
one) As I mentioned in an email.. the
Government, Military, Civilian agencies, private entities, local/state
government have ALL publicly stated the Amateur Radio is a very valuable
resource to them and they publicly support its retention!

73

Chuck K3FT
k3ft@erols.com

I am not an ARRL member.

9. Re: [HFSIG:1940] Re: The ARRL Letter Vol 16, No 5

From: Chuck K3FT <k3ft@erols.com>
Date: Sun, 02 Feb 1997 17:39:04 -0500
To: hfsig@tapr.org
Message-ID: <32F51788.4043@erols.com>
Rendered body type: text/plain

Thomas Ringate wrote:
>

> My opinion is that as a responsible citizen of the USA, it's time to
> abolish amateur radio, and give the freq. back to the citizens of the
> USA. There are over 300 million citizens of the USA, but only a handful
> receive any use of this precious asset. How can any of us look at our
> neighbors and say we are making good use of an asset worth billions of
> $$ to the USA, and we abuse it like we do.

(Response clipped due to Bandwidth Reduction Act of 1996)

Thomas, the statements you make are patently without merit and have
absolutely no foundation inn truth. Please take a minute to research the
benefits that Amateur Radio has (and still does) provided to the public
since its' inception. (any decent library can help you)

I think, after reflection and research, you might think differently

Chuck K3FT
k3ft@erols.com

10. Re: The ARRL Letter Vol 16, No 5

From: "James D. Mayfield" <jdm326@revealed.net>
Date: Sun, 02 Feb 1997 17:03:49 -0600
To: Thomas Ringate <ThomasLRingate@ij.net>, larryj@cetlink.net
Cc: KB6ZVP@aol.com, elindquist@arrl.org, amsat-bb@amsat.org, netsig@tapr.org,
        tapr-bb@tapr.org, hfsig@tapr.org, bbssig@tapr.org, fbb@f6fbb.org,
        rick@janrix.com, onemm@cetlink.net
Message-ID: <3.0.32.19970202170337.006b85a4@revealed.net>
Rendered body type: text/plain

What a nut!!!

At 04:17 PM 2/2/97 -0500, Thomas Ringate wrote:
>Larry W. Johnson wrote:
>

>>   Just remember opinions are the most common thing around,everyone has
>> one but every one forunately does not have to share the same one.
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>jokers who are using it.
>
>Oh yes, I have also been an amateur since 1957, but I have been a
>citizen of the USA since 1944 and somehow that responsibility weighs
>more heavily than the simple enjoyment I get from talking on the radio
>every now and then.
>
>Keep CW, in fact make it 50 words per minute for the novice. Let's
>get this over with as fast as we can so all 300+ million of us can get
>some use out of this asset.
>
>If we each had to pay a license fee to compensate for the lost
>revenue this spectrum is worth, Bill Gates would be the only person
>who could afford a license.
>
>My vote is to end amateur radio and make a valid use of this
>precious spectrum for the good of all, not just a few old fuddy duddies.
>

-               Best '73   -
-           J. Dave Mayfield   -
-   http://home.revealed.net/qste/bnr   -
-           qste@revealed.net   -
-              309-797-3827   -
----

11. Re: The ARRL Letter Vol 16, No 5

From: Nicolaus Sallay <nsallay@internext.com.br>
Date: Sun, 02 Feb 1997 22:00:44 -0400
To: Thomas Ringate <ThomasLRingate@ij.net>
Cc: larryj@cetlink.net, KB6ZVP@aol.com, elindquist@arrl.org,
        amsat-bb@amsat.org, netsig@tapr.org, tapr-bb@tapr.org, hfsig@tapr.org,
        bbssig@tapr.org, fbb@f6fbb.org, rick@janrix.com, onemm@cetlink.net
Message-ID: <32F54659.53A9@internext.com.br>
Rendered body type: text/plain

Hello Sir,
Excuse me,it`s long time ago,that I heard such a amount of nonsense
written down in a specific place like this.Stand by,please and reflect
only a few moments about your sentences.
Best regards and 73
Nicolaus K.S.Sallay - PP8DA
PP8DA@amsat.org
nsallay@internext.com.br
==========
Thomas Ringate wrote:
>
> Larry W. Johnson wrote:
>
> >   Just remember opinions are the most common thing around,everyone has
> > one but every one forunately does not have to share the same one.
>
> My opinion is that as a responsible citizen of the USA, it's time to
> abolish amateur radio, and give the freq. back to the citizens of the
> USA. There are over 300 million citizens of the USA, but only a handful
> receive any use of this precious asset. How can any of us look at our
> neighbors and say we are making good use of an asset worth billions of
> $$ to the USA, and we abuse it like we do.
>
> We should all be ashamed. We do not provide a service that is worth
> the cost amateur radio is costing the rest of the non-amateur population

> of this country. If the facts were presented in a national election,
> and this were put to a vote of the general poplualtion, amateur radio
> would be history never again to be heard from.
>
> There is no justification to reserve a billion dollar asset for the
> jokers who are using it.
>
> Oh yes, I have also been an amateur since 1957, but I have been a
> citizen of the USA since 1944 and somehow that responsibility weighs
> more heavily than the simple enjoyment I get from talking on the radio
> every now and then.
>
> Keep CW, in fact make it 50 words per minute for the novice. Let's
> get this over with as fast as we can so all 300+ million of us can get
> some use out of this asset.
>
> If we each had to pay a license fee to compensate for the lost
> revenue this spectrum is worth, Bill Gates would be the only person
> who could afford a license.
>
> My vote is to end amateur radio and make a valid use of this
> precious spectrum for the good of all, not just a few old fuddy duddies.

12. [HFSIG:1940] Re: The ARRL Letter Vol 16, No 5

From: kl7ev@actis.polarnet.com
Date: 2 Feb 1997 17:15:54 -0800
To: hfsig@tapr.org
Message-ID: <199702030214.UAA23568@tapr.org>
Rendered body type: text/plain

HF>Larry W. Johnson wrote:

HF>>   Just remember opinions are the most common thing around,everyone has
...

HF>My opinion is that as a responsible citizen of the USA, it's time to
abolish amateur radio, ...

HF>We should all be ashamed. We do not provide a service that is worth
...

HF>There is no justification to reserve a billion dollar asset for the
HF>jokers who are using it.

HF>Oh yes, I have also been an amateur since 1957, but I have been a
HF>citizen of the USA since 1944 ...

HF>Keep CW, in fact make it 50 words per minute for the novice. Let's
...

HF>If we each had to pay a license fee to compensate for the lost
HF>revenue ...

HF>My vote is to end amateur radio and make a valid use of this
HF>precious spectrum for the good of all, not just a few old fuddy duddies.

In my opinion, people should be allowed to state their opinion only when
they are willing to identify themselves and thus take any heated
comments might come as a result of their opinion. We don't need this
Back Stabbing and I for one would like to see people sign their messages
so people like me don't have to respond publicly to an anonomous noise
maker who doesn't have the decency (sp) to post their return e-mail
address! I would prefer to keep my comments private but this individual
chooses to hide behind their statements without giving the other side of
the story an opportunity to respond. Regards,

       - Greg -
---
kl7ev@actis.polarnet.com
-

13. Re: The ARRL Letter Vol 16, No 5

From: k5cnf1@juno.com (Richard B Morrow)
Date: Sun, 2 Feb 1997 20:34:18 PST
To: hfsig@tapr.org
Message-ID: <19970202.204156.4535.0.k5cnf1@juno.com>
Rendered body type: text/plain

You have hit it on the head. It is a beneficial hobby for those of us
that enjoy it and there are plenty of advances to be made, despite what
some people believe. I am not much of a cw op. but it was and still is
used in a lot of places, for many reasons.

I also am not an ARRL member, nor am I an old fuddy duddy, and I do
remember the uproar when ssb kicked am in the rump and took over the
bands. People seem to over look the fact that satcom can be killed both
by nature and by hostile forces. Cw has its' place, just lke drag racing
and things of that sort. Those that does it likes it, them that don't
,don't

73, K5CNF Richard Morrow, licensed since 1955

14. Re: The ARRL Letter Vol 16, No 5

From: k5cnf1@juno.com (Richard B Morrow)
Date: Sun, 2 Feb 1997 20:41:52 PST
To: hfsig@tapr.org
Message-ID: <19970202.204156.4535.1.k5cnf1@juno.com>
Rendered body type: text/plain

>> The only organization on the planet stuck in this rut is the
>League.(de k3ft NOT TRUE. Many governments, private entities, and
>individuals
>still use CW. Yes, it is old and outdated (in terms of throughput and
>bulk-handling abilities) but still a viable communications medium.

And it does get the job done under adverse conditiions with simple rigs.

>The military DOES still teach CW for certain types of military ops.

Special Forces comes to mind

>>   Why not test on one's ability to operate voice? .. packet? ..
>ATV?de k3ft - how about operational testing? proper ettiquette,
>procedure,
I know a LOT of people who would flunk that last part! some of them are
Xtras too.

>
>de k3ft - testing required by ITU regs. they test on other modes.
>
>>   The code is dead, (NOPE... OLD AND OUTDATED.. not the same thing)
>and Amateur radio will die if you/we do not advance the
>> science and the membership. (de k3ft - being done as we speak. by
>many groups - TAPR with APRS for one) As I mentioned in an email.. the

Right! old and maybe outdated, but still useful...

>Government, Military, Civilian agencies, private entities, local/state
>
>government have ALL publicly stated the Amateur Radio is a very
>valuable
>resource to them and they publicly support its retention!
>
>73
>
>Chuck K3FT
>k3ft@erols.com
>
>I am not an ARRL member.

Neither am I.

73, from K5CNF, Richard Morrow

15. Re: [HFSIG:1945] Re: The ARRL Letter Vol 16, No 5

From: "Stan LaPointe" <lapointe@hopf.dnai.com>
Date: Sun, 2 Feb 1997 20:12:54 -0800
To: <hfsig@tapr.org>
Message-ID: <199702030411.UAA10752@dnai.com>
Rendered body type: text/plain

And please, check with the pharmacy to see if your Prozac refill is ready
yet.

> From: Nicolaus Sallay <nsallay@internext.com.br>
> To: hfsig@tapr.org

> Subject: [HFSIG:1945] Re: The ARRL Letter Vol 16, No 5
> Date: Sunday, February 02, 1997 6:06 PM
>
> Hello Sir,
> Excuse me,it`s long time ago,that I heard such a amount of nonsense
> written down in a specific place like this.Stand by,please and reflect
> only a few moments about your sentences.
> Best regards and 73
> Nicolaus K.S.Sallay - PP8DA
> PP8DA@amsat.org
> nsallay@internext.com.br
> ==
> Thomas Ringate wrote:
> >
> > Larry W. Johnson wrote:
> >
> > > >   Just remember opinions are the most common thing around,everyone
has
> > > one but every one forunately does not have to share the same one.
> >

16. Re: The ARRL Letter Vol 16, No 5

From: Radio Amateurs Assembly - RAA <radioass@idsc.gov.eg>
Date: Mon, 3 Feb 1997 06:23:25 +0000 (africa)
To: Thomas Ringate <ThomasLRingate@ij.net>
Cc: larryj@cetlink.net, KB6ZVP@aol.com, elindquist@arrl.org,
amsat-bb@amsat.org, netsig@tapr.org, tapr-bb@tapr.org, hfsig@tapr.org,
bbssig@tapr.org, fbb@f6fbb.org, rick@janrix.com, onemm@cetlink.net
Message-ID: <Pine.SOL.3.94.970203061105.4102D-100000@idsc.gov.eg>
Rendered body type: text/plain

Hi folks!

What is this guy saying!

First it seemed to be a "national" war between fellows of the ARRL on the code issue. On the sudden, he came with this ingenious idea of selling the amateur spectrum and abolishing amateur radio!

What a joke! He forgot that the amateur spectrum is not a property of the USA or any other single country to ask for "presenting it in an national election"! It is owned by all the international community. This guy is not a radio amateur and never been an amateur one day. Being an amateur is not just a piece of paper to hang on the wall...I even doubt he ever had one! Radio amateurs are the people who made the radio spectrum worth these billions of dollars he is talking about and they will continue to add value to it whether he likes it or not.

I noticed that after the Internet has become that popular...a lot of simple minded half-knowledged and let me say "untrue" amateurs fall in the dilemma of asking themselves "if it is so easy and funny to use the internet to communicate with other people..why to go thru all the fuss of amateur radio?". They ignore that the Internet as well as all the modern communications avenues are merely branches on the road paved by amateur radio and radio amateurs during their history. In other words all are coming out from the cloak of amateur radio. Anyway, it needs to be a radio amateur first to understand that amateur radio is not the "simple enjoyment ..of talking on the radio".

Hossam El Shenawy, SU1HM
Network Manager,
Egypt Radio Amateurs Assembly

email(alias): SU1HM@amsat.or

17. Re: The ARRL Letter Vol 16, No 5

From: "Larry W. Johnson" <larryj@cetlink.net>
Date: Wed, 15 Jan 1997 23:53:42 -0500
To: "Radio Amateurs Assembly - RAA" <radioass@idsc.gov.eg>, <SU1HM@amsat.org>
Cc: <KB6ZVP@aol.com>, <onemm@cetlink.net>, <elindquist@arrl.org>,
<amsat-bb@amsat.org>, <netsig@tapr.org>, <tapr-bb@tapr.org>,
<hfsig@tapr.org>, <bbssig@tapr.org>, <rick@janrix.com>,
<qste@revealed.net>, <PP8DA@amsat.org>, <ThomasLRingate@.ij.net>
Message-ID: <199702030504.AAA05020@moltar.cetlink.net>
Rendered body type: text/plain

Hello Hossam,

  Unfortunately this guy appears to have a current ham ticket, he did not sign his Call but as near as I can figure He is K0ZXF currently living in Tampa,Florida.

  I can't believe he has been a Ham for that long and has always had those feelings about the hobby.If he has I guess we can consider a true hippocrite.

  I wish to thank all of you that has sent a copy of your responces to me but please delete fbb@f6fbb.org from any further responces.Per Olsen has requested this.

73's to all
Larry Johnson - W7LJ

> From: Radio Amateurs Assembly - RAA <radioass@idsc.gov.eg>
> To: Thomas Ringate <ThomasLRingate@ij.net>
> Cc: larryj@cetlink.net; KB6ZVP@aol.com; elindquist@arrl.org; amsat-bb@amsat.org; netsig@tapr.org; tapr-bb@tapr.org; hfsig@tapr.org; bbssig@tapr.org; fbb@f6fbb.org; rick@janrix.com; onemm@cetlink.net
> Subject: Re: The ARRL Letter Vol 16, No 5
> Date: Monday, February 03, 1997 1:23 AM
>
> Hi folks!
>
> What is this guy saying!
>
> First it seemed to be a "national" war between fellows of the ARRL on the
> code issue. On the sudden, he came with this ingenious idea of selling the
> amateur spectrum and abolishing amateur radio!
>
> What a joke! He forgot that the amateur spectrum is not a property of the

> USA or any other single country to ask for "presenting it in an national
> election"! It is owned by all the international community. This guy is not
> a radio amateur and never been an amateur one day. Being an amateur is not
> just a piece of paper to hang on the wall...I even doubt he ever had one!

> Radio amateurs are the people who made the radio spectrum worth these
> billions of dollars he is talking about and they will continue to add
> value to it whether he likes it or not.
>
>
> I noticed that after the Internet has become that popular...a lot of
> simple minded half-knowledged and let me say "untrue" amateurs fall in the
> dilemma of asking themselves "if it is so easy and funny to use the internet to
> communicate with other people..why to go thru all the fuss of amateur radio?".
> They ignore that the Internet as well as all the modern communications avenues
> are merely branches on the road paved by amateur radio and radio amateurs

> during their history. In other words all are coming out from the cloak of
> amateur radio. Anyway, it needs to be a radio amateur first to understand

> that amateur radio is not the "simple enjoyment ..of talking on the radio".
>
> ------
>
> Hossam El Shenawy, SU1HM
> Network Manager,
> Egypt Radio Amateurs Assembly
>
> email(alias): SU1HM@amsat.or
>
>

18. [HFSIG:1949] Re: The ARRL Letter Vol 16, No 5

From: kl7ev@actis.polarnet.com
Date: 2 Feb 1997 21:23:30 AST
To: hfsig@tapr.org
Message-ID: <199702030622.AAA18704@tapr.org>
Rendered body type: text/plain

I happen to agree with mr Nicolaus Salay, so does that mean I need to
check for a prescription of Prozac as well!?!??!

ps.

Are you to CHICKEN to post your e-mail return address?

       - Greg -
---
kl7ev@actis.polarnet.com

HF>And please, check with the pharmacy to see if your Prozac refill is ready
HF>yet.
HF>----------
HF>> From: Nicolaus Sallay <nsallay@internext.com.br>
HF>> To: hfsig@tapr.org
HF>> Subject: [HFSIG:1945] Re: The ARRL Letter Vol 16, No 5
HF>> Date: Sunday, February 02, 1997 6:06 PM
HF>>
HF>> Hello Sir,
HF>> Excuse me,it`s long time ago,that I heard such a amount of nonsense
HF>> written down in a specific place like this.Stand by,please and reflect
HF>> only a few moments about your sentences.
HF>> Best regards and 73
HF>> Nicolaus K.S.Sallay - PP8DA
HF>> PP8DA@amsat.org
HF>> nsallay@internext.com.br
HF>> ==========
HF>> Thomas Ringate wrote:
HF>> >
HF>> > Larry W. Johnson wrote:
HF>> >
HF>> > >   Just remember opinions are the most common thing around,everyone
HF>has
HF>> > > one but every one forunately does not have to share the same one.
HF>> >

19. Re: The ARRL Letter Vol 16, No 5

From: "Jean-Paul ROUBELAT" <jpr@f6fbb.org>
Date: Mon, 3 Feb 1997 07:21:20 +0100
To: "Radio Amateurs Assembly - RAA" <radioass@idsc.gov.eg>,
        "Thomas Ringate" <ThomasLRingate@ij.net>
Cc: <larryj@cetlink.net>, <KB6ZVP@aol.com>, <elindquist@arrl.org>,
        <amsat-bb@amsat.org>, <netsig@tapr.org>, <tapr-bb@tapr.org>,
        <hfsig@tapr.org>, <bbssig@tapr.org>, <fbb@f6fbb.org>,
        <rick@janrix.com>, <onemm@cetlink.net>
Message-ID: <199702030702.IAA06404@aris1.eb3ago.org>
Rendered body type: text/plain

Hi All,

I'm sorry but fbb mailing list is devoted to FBB technical usage. This list
is not well adapted to your debate.

Please remove the address fbb@f6fbb.org from your list.

Thank-you.

_____

73s

Jean-Paul ROUBELAT, jpr@f6fbb.org

> De : Radio Amateurs Assembly - RAA <radioass@idsc.gov.eg>
> A : Thomas Ringate <ThomasLRingate@ij.net>
> Cc : larryj@cetlink.net; KB6ZVP@aol.com; elindquist@arrl.org;
amsat-bb@amsat.org; netsig@tapr.org; tapr-bb@tapr.org; hfsig@tapr.org;
bbssig@tapr.org; fbb@f6fbb.org; rick@janrix.com; onemm@cetlink.net
> Objet : Re: The ARRL Letter Vol 16, No 5
> Date : lundi 3 février 1997 07:23
>

> Hi folks!
>
> What is this guy saying!
>
> First it seemed to be a "national" war between fellows of the ARRL on the
> code issue. On the sudden, he came with this ingenious idea of selling
the
> amateur spectrum and abolishing amateur radio!
>
> What a joke! He forgot that the amateur spectrum is not a property of the

> USA or any other single country to ask for "presenting it in an national
> election"! It is owned by all the international community. This guy is
not
> a radio amateur and never been an amateur one day. Being an amateur is
not
> just a piece of paper to hang on the wall...I even doubt he ever had one!

> Radio amateurs are the people who made the radio spectrum worth these
> billions of dollars he is talking about and they will continue to add
> value to it whether he likes it or not.
>
>
> I noticed that after the Internet has become that popular...a lot of
> simple minded half-knowledged and let me say "untrue" amateurs fall in
the
> dilemma of asking themselves "if it is so easy and funny to use the
internet to
> communicate with other people..why to go thru all the fuss of amateur
radio?".
> They ignore that the Internet as well as all the modern communications
avenues
> are merely branches on the road paved by amateur radio and radio amateurs

> during their history. In other words all are coming out from the cloak of
> amateur radio. Anyway, it needs to be a radio amateur first to understand

> that amateur radio is not the "simple enjoyment ..of talking on the
radio".
>
> ------
>
> Hossam El Shenawy, SU1HM
> Network Manager,
> Egypt Radio Amateurs Assembly
>
> email(alias): SU1HM@amsat.or
>

20. Re: [HFSIG:1940] Re: The ARRL Letter Vol 16, No 5

From: sams@jetlink.net (Sam Stanifer)
Date: Sun, 2 Feb 1997 23:30:58 +0800
To: hfsig@tapr.org
Message-ID: <v01530500af1a5c07c48e@[206.72.64.179]>
Rendered body type: text/plain

Thomas Ringate wrote (tongue-in-cheek?):

 There are over 300 million citizens of the USA, but only a handful
>receive any use of this precious asset.

Most of these 300 million "benefit" from the endless drivel called
television, which occupies many times more exclusive spectrum than amateur
radio and probably most any other single service.

(sigh) 73, KJ6BU

21. Re: [HFSIG:1936] Re: The ARRL Letter Vol 16, No 5

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Mon, 3 Feb 1997 07:33:26 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.970203073140.9146A-100000@www.kelly-afb.org>
Rendered body type: text/plain

I like CW because its _fun_.

I would like to see 4800 BPS of robust data signals on the HF hambands as
well as SS...but I would still want to work CW...its just fun and relaxing.

Walt

```
    ========================================================================
    | The Micro$oft operating system didn't get as bad as it is overnight,|
    | it has taken over 10 years of careful, calculated development.      |
    ========================================================================
    |                                  |                                   |
    |                                  | The greatest dangers to liberty   |
    | Walt DuBose - K5YFW              | lurk in insidious encroachment    |
    | E-Mail k5yfw@www.kelly-afb.org   | by men of zeal, well-meaning      |
    | Business Telephone: (210)925-1171 | but without understanding.       |
    |     Home Telephone: (210)696-3196 |                                  |
    |                                  |   - Justice Louis D. Brandeis     |
    |                                  |                                   |
    ========================================================================
```

22. Re: [HFSIG:1950] Re: The ARRL Letter Vol 16, No 5

From: "Peter Markham" <pmarkham@up.net>
Date: Mon, 3 Feb 1997 08:58:48 -0500
To: hfsig@tapr.org
Message-ID: <9702031357.AA27799@btc1>
Rendered body type: text/plain

Date: Sun, 2 Feb 1997 22:33:59 -0600 (CST)
Reply-to: hfsig@tapr.org
From: Radio Amateurs Assembly - RAA <radioass@idsc.gov.eg>
To: hfsig@tapr.org
Subject: [HFSIG:1950] Re: The ARRL Letter Vol 16, No 5

Hossam said:

    Anyway, it needs to be a radio amateur first to understand
that amateur radio is not the "simple enjoyment ..of talking on the radio".

Pete says:

    That is what has been lost in the democratisation and marketing
of amateur radio, in this country; a price for our success (or lack
thereof?).

Pete/wa4hei

The meek will not inherit the Earth.

23. Re: [HFSIG:1955] Re: The ARRL Letter Vol 16, No 5

From: "Charles Hoffman" <choffman@pelican.davlin.net>
Date: Mon, 3 Feb 1997 10:47:43 -0600
To: <hfsig@tapr.org>
Message-ID: <199702031655.KAA09595@pelican.davlin.net>
Rendered body type: text/plain

Howdy! I noticed that the list owner killed the thread...also noted that
Richard Morrow K5CNF was stirred out of lurkingville onto the thread. My
comment on cw: It is the secret handshake of the amateur radio
fraternity; it does not require justification. Anyone who does not or
cannot give the handshake is not a fraternal member. Licensed maybe, but
not within the inner-circle.

```
----------
| From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
| To: hfsig@tapr.org
| Subject: [HFSIG:1955] Re: The ARRL Letter Vol 16, No 5
| Date: Monday, February 03, 1997 07:36
|
| I like CW because its _fun_.
|
| I would like to see 4800 BPS of robust data signals on the HF hambands as

| well as SS...but I would still want to work CW...its just fun and
relaxing.
|
| Walt
|
|
========================================================================
|     | The Micro$oft operating system didn't get as bad as it is
overnight,|
|     | it has taken over 10 years of careful, calculated development.
```

```
|
|
=======================================================================
|     |                                   |
|
|     |                                   | The greatest dangers to liberty
|
|     | Walt DuBose - K5YFW               | lurk in insidious encroachment
|
|     | E-Mail k5yfw@www.kelly-afb.org    | by men of zeal, well-meaning
|
|     | Business Telephone: (210)925-1171 | but without understanding.
|
|     |     Home Telephone: (210)696-3196 |                               |
|     |                                   |   - Justice Louis D. Brandeis
|
|     |                                  |                               |
|
=======================================================================
|
```

24. Discuss CW elsewhewre !

```
From: Iztok Saje <s52d@s55tcp.ampr.org>
Date: Mon, 03 Feb 97 17:46:24 EST
To: hfsig@tapr.org
Message-ID: <1885@s55tcp.ampr.org>
Rendered body type: text/plain
```

If all anti-CW proponents would spend their time learning CW instead
of spoiling this (and other) conferences with they crying:
"Mammy, mammy, I wanna 20m ! Gimme 14.005 !"
we all would be much more happy.
So please discuss it where it is place to discuss !

And beside spending some 200 to 300 hours on the band to get some
practice and pass CW exam: Even good English is not needed to pass
FCC exams. Just optical memory of how Q and A look is enough for Extra.

Proof:
      73 gl,
            Iztok, WU2D
p.s. I will not repeat NOT continue any such discussions here. Flames to
s52d@ljutcp.hamradio.si, not to this conference.
..-.-

25. Re: [HFSIG:1957] Re: The ARRL Letter Vol 16, No 5

```
From: George Cserenyi <zsolt@direct.ca>
Date: Mon, 3 Feb 1997 11:40:01 -0800 (PST)
To: Charles Hoffman <choffman@pelican.davlin.net>
Cc: hfsig@tapr.org
Message-ID: <Pine.LNX.3.95.970203113720.11995C-100000@cirmos.ci.ca>
Rendered body type: text/plain
```

Charles, how did you do that?
This is the first time, my pine mailer can't read a message.
Any suggestion by other pine users?

On Mon, 3 Feb 1997, Charles Hoffman wrote:

```
Date: Mon, 3 Feb 1997 11:00:37 -0600 (CST)
From: Charles Hoffman <choffman@pelican.davlin.net>
Reply-To: hfsig@tapr.org
To: hfsig@tapr.org
Subject: [HFSIG:1957] Re: The ARRL Letter Vol 16, No 5
Parts/attachments:
   1   OK    46 lines  Text (charset: ISO-8859-1)
   2        ~64 lines  Text (charset: ISO-8859-1)
----------------------------------------


  [Part 1, Text/PLAIN (charset: ISO-8859-1 "Latin 1")  46 lines]
  [Not Shown. Use the "V" command to view or save this part]


  [Part 2, Text/HTML (charset: ISO-8859-1 "Latin 1")  64 lines]
  [Cannot display this part. Press "V" then "S" to save in a file]
```

George

26. Apologize!

From: "Emese Tolnai" <temese@hotmail.com>
Date: Mon, 3 Feb 1997 11:21:59 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199702031921.LAA21767@f3.hotmail.com>
Rendered body type: text/plain

All those who has been abusing this list to promote their
religion, please stop it and apologize.
Here is an example from another list, if you don't know how.

Subject: Apology

I would like to give an apology to the members of the FBB mail list.

When I responded to Larry's original email with a very controversal
opinion concerning the future of amateur radio in the USA I did not
properly review all the lists Larry's note was sent to.

That post has resulted in many posts to the FBB list that have
nothing at all to do with the development of FBB.

Please accept my apology for not properly checking all the lists
my post was going to.

I am a member of the FBB list, and have found many useful suggestions
in my operation of FBB. I did not intend to bring this other
debate to this list.

Please accept my apology, not for having an opinion, but for posting it
to a list where it did not belong.

Get Your *Web-Based* Free Email at http://www.hotmail.com
---

27. RE: [HFSIG:1958] Discuss CW elsewhewre !

From: "Borcher, David (NG-MS)" <DBORCHER@nogmail.gi.com>
Date: Mon, 03 Feb 1997 13:20:00 -0800 (PST)
To: hfsig <hfsig@tapr.org>
Message-ID: <32F64879@gismtpgate.gi.com>
Rendered body type: text/plain

I second the motion...

Dave K7NG

From: hfsig
To: hfsig
Subject: [HFSIG:1958] Discuss CW elsewhewre !
Date: Monday, February 03, 1997 12:00PM

If all anti-CW proponents would spend their time learning CW instead
of spoiling this (and other) conferences with they crying:
"Mammy, mammy, I wanna 20m ! Gimme 14.005 !"
we all would be much more happy.
So please discuss it where it is place to discuss !

And beside spending some 200 to 300 hours on the band to get some
practice and pass CW exam: Even good English is not needed to pass
FCC exams. Just optical memory of how Q and A look is enough for Extra.

Proof:
      73 gl,
              Iztok, WU2D
p.s. I will not repeat NOT continue any such discussions here. Flames to
s52d@ljutcp.hamradio.si, not to this conference.
.-.-

28. snr

From: Glen Worstell <Glen_Worstell@notes.seagate.com>
Date: 3 Feb 97 14:26:12
To: hfsig <hfsig@tapr.org>
Message-ID: <9702032022.AA0247@notes.seagate.com>
Rendered body type: text/plain

The SNR of this group has gone negative. Too bad, it used
to be pretty interesting.

Would someone please tell me how to unsubscribe?

TIA,
Glen.

29. SSTV demod

From: "Wheatley, Maurice" <mwheatle@trsystems.com>
Date: Mon, 3 Feb 1997 10:40:10 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=TRSystems%l=WEBSERVER2-970203154010Z-164@www.trsystems.com>
Rendered body type: text/plain

Hi,
I've been playing around with audio FM detection of SSTV signals using
DSP. What would be the best method for demodulation? I've tried the
simple limiter/discriminator which works but seems to underutilize the
advantages of DSP. Initial attempts of DPLL's have resulted in too
narrow of a bandwidth and poor dynamic range. I'm now implementing the
non-limiting scheme using arctan(I/Q) followed by a differentiator and
correction filter described in Marvin Frerking's book.   Any suggestions
before I go down some deadend path?

Thanks.

Moe Wheatley AE4JY
mwheatle@trsystems.com

30. Re: [HFSIG:1957] Re: The ARRL Letter Vol 16, No 5

From: mwestfal@csci.csusb.edu (Michael Westfall)
Date: Mon, 3 Feb 1997 17:30:05 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199702040130.BAA05427@csci.csusb.edu>
Rendered body type: text/plain

In a former treatise, Charles Hoffman spake thusly:
->
-> This is a multi-part message in MIME format.
->
-> ------=_NextPart_000_01BC11BF.AF49B400
-> Content-Type: text/plain; charset=ISO-8859-1
-> Content-Transfer-Encoding: 7bit
->
-> Howdy! I noticed that the list owner killed the thread...also noted that
-> Richard Morrow K5CNF was stirred out of lurkingville onto the thread. My
-> comment on cw: It is the secret handshake of the amateur radio
-> fraternity; it does not require justification. Anyone who does not or
-> cannot give the handshake is not a fraternal member. Licensed maybe, but
-> not within the inner-circle.
->

Oh, brother!

I bet you're a member of The Lodge, too....

The Amateur Radio Illuminati. Yeah, that's it...

---
73 de Mike,   ax.25net: N6KUY@W6JBT.#SOCA.CA.USA.NOAM
                  amprnet: n6kuy@n6kuy.ampr.org [44.18.0.49]
                 internet: mwestfal@.csci.csusb.edu
                      web: http://orion.csci.csusb.edu:8080
Linux: the Gates of Hell shall not prevail.
GCS/M { -d+ p+ c++ l u++ e+(*) m++(-) s/+ !n-(---) h-- !f g+ w+ t++ r-(--) y+ }
---

31. Re: [HFSIG:1964] Re: The ARRL Letter Vol 16, No 5

From: "Charles Hoffman" <choffman@pelican.davlin.net>
Date: Mon, 3 Feb 1997 21:14:20 -0600
To: <hfsig@tapr.org>
Message-ID: <199702040322.VAA18122@pelican.davlin.net>
Rendered body type: text/plain

I suppose I should respond to clear up a misunderstanding; don't take my
post so seriously. I meant it to go email to a radio pal of almost 40
years with whom there is a private bit of humor which requires no
explanation. Sorry for the consternation...just disregard as cockpit error
OM.

A lodge member, yes, the Order of the Arrow, BSA, circa 1954 roughly. A
private comment not intended for the public, so disregard it as such if you
would be so kind. Just be glad you were not on the airplane.

73

| From: Michael Westfall <mwestfal@csci.csusb.edu>
|My
| -> comment on cw: It is the secret handshake of the amateur radio
| -> fraternity; it does not require justification. Anyone who does not or
| -> cannot give the handshake is not a fraternal member. Licensed maybe,
but
| -> not within the inner-circle.
| ->
|
|
| Oh, brother!
|
| I bet you're a member of The Lodge, too....
|
| The Amateur Radio Illuminati. Yeah, that's it...
|
|
|

| 73 de Mike,   ax.25net: N6KUY@W6JBT.#SOCA.CA.USA.NOAM
| amprnet: n6kuy@n6kuy.ampr.org [44.18.0.49]
| internet: mwestfal@.csci.csusb.edu
| web: http://orion.csci.csusb.edu:8080
| Linux: the Gates of Hell shall not prevail.
| GCS/M { -d+ p+ c++ l u++ e+(*) m++(-) s/+ !n-(---) h-- !f g+ w+ t++
r-(--) y+ }
|

|

32. Re: [HFSIG:1965] Re: The ARRL Letter Vol 16, No 5

From: George Cserenyi <zsolt@direct.ca>
Date: Mon, 3 Feb 1997 21:33:40 -0800 (PST)
To: Charles Hoffman <choffman@pelican.davlin.net>
Cc: hfsig@tapr.org
Message-ID: <Pine.LNX.3.95.970203212903.13664B-100000@cirmos.ci.ca>
Rendered body type: text/plain

Hello Charles,
                what kind of a mailer are you using?
I've sent you 2 private notes, but haven't seen any replies.
You are the only one on this list who send this kind of mail.
I apologise to the list, but wanted to alert Charles.
George

On Mon, 3 Feb 1997, Charles Hoffman wrote:

<html><head></head><BODY bgcolor="#FFFFFF"><p>I suppose I
should respond to clear up a misunderstanding; don't take my post so seriously. I meant it
to go email to a radio pal of almost 40 years with w
hom there is a private bit of humor which requires no explanation. Sorry for the
consternation...just disregard as cockpit error OM.

A lodge member, yes, the Order of the Arrow,
 BSA, circa 1954 roughly. A private comment not intended f
or the public, so disregard it as such if you would be so kind. Just be glad you were not on the
airplane.

73

| From: Michael
Westfall <<u>mwestfal@csci.csusb.
edu</u>>
|My
| -> comment on cw: It is the secret
handshake of the amateur radio
| -> fraternity; it does not require justification. Anyone
who does not or
| -> cannot give the handshake is n
ot a fraternal member. Licensed maybe, but
| -> not within the inner-circle.
|
->
|
|
| Oh, brother!
|
| I bet you're a member of The Lodge, too....
|
|
The Amateur Radio Illuminati. Yeah, that's it...
|
|
| ----------
| 73
de Mike,   ax.25net: <u>N6KUY@W6JBT.</u>#SOCA.CA.USA.NOAM
|
 amprnet: <u>n6kuy@n6kuy.ampr.org</u>
[44.18.0.49]
| &nb
sp;   internet:
   <u>mwestfal@.csci.csusb.edu</u>
|

;   web: <u>http://orion.csci.csusb.edu:8080</u>
| Linux: the Gates of
Hell shall not prevail.
| GCS/M { -d+ p+ c++ l u++ e+(*) m+
+(-) s/+ !n-(---) h-- !f g+ w+ t++ r-(--) y+ }
|

| </p>
</body></html>

33. Re: The ARRL Letter Vol 16, No 5

From: k5cnf1@juno.com (Richard B Morrow)
Date: Mon, 3 Feb 1997 22:49:39 PST
To: hfsig@tapr.org
Message-ID: <19970203.224957.4535.1.k5cnf1@juno.com>
Rendered body type: text/plain

>
>Howdy! I noticed that the list owner killed the thread...also noted
>that
>Richard Morrow K5CNF was stirred out of lurkingville onto the thread.

(I am always lurking some where, K5CNF)

>My
>comment on cw: It is the secret handshake of the amateur radio
>fraternity; it does not require justification. Anyone who does not or
>cannot give the handshake is not a fraternal member. Licensed maybe,
>but
>not within the inner-circle.

CW is still used in the marine world, anyone who doubts this, listen to
the 4,6,8,12,18,and 22 MHz marine cw frequencies, there is still plenty
of cw flying around...That is the last I am going to say about it also
and am going to crawl back into the noise level until disturbed again

Hi Walt, Wondered where you were.

73, RM, K5CNF

34. Re: [HFSIG:1963] SSTV demod

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Tue, 04 Feb 1997 00:14:09 -0500
To: hfsig@tapr.org
Message-ID: <199702040514.AAA11154@aus-e.mp.campus.mci.net>
Rendered body type: text/plain

At 03:27 PM 2/3/97 -0600, you wrote:
>Hi,
>I've been playing around with audio FM detection of SSTV signals using
>DSP. What would be the best method for demodulation? I've tried the
>simple limiter/discriminator which works but seems to underutilize the
>advantages of DSP. Initial attempts of DPLL's have resulted in too
>narrow of a bandwidth and poor dynamic range. I'm now implementing the
>non-limiting scheme using arctan(I/Q) followed by a differentiator and
>correction filter described in Marvin Frerking's book.   Any suggestions
>before I go down some deadend path?
>
>Thanks.
>
>Moe Wheatley AE4JY
>mwheatle@trsystems.com
>

Moe,

I was going to suggest a differentiating filter but I guess you are "on" to
that. Could you post the title/publisher of Frekring's book?

73's
Don

35. Re: [HFSIG:1963] SSTV demod

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Tue, 04 Feb 1997 08:52:44 +0100
To: hfsig@tapr.org
Message-ID: <32F6EACC.683F@ife.ee.ethz.ch>
Rendered body type: text/plain

> narrow of a bandwidth and poor dynamic range. I'm now implementing the
> non-limiting scheme using arctan(I/Q) followed by a differentiator and
> correction filter described in Marvin Frerking's book.   Any suggestions
> before I go down some deadend path?
I've used this successfully in my HF FAX demodulator...
look at http://www.ife.ee.ethz.ch/~sailer/ham/linux/fax.html

> Moe Wheatley AE4JY
Tom, AE4WA (HB9JNX)

---

36. RE: [HFSIG:1968] Re: SSTV demod

From: "Wheatley, Maurice" <mwheatle@trsystems.com>
Date: Tue, 4 Feb 1997 09:13:52 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=TRSystems%l=WEBSERVER2-970204141352Z-236@www.trsystems.com>
Rendered body type: text/plain

Don,
The book is "Digital Signal Processing in Communication Systems" by
Marvin E. Frerking ISBN 0-442-01616-6.
It seems to have a pretty good balance of theory to practical
algorithms.

>
>>Moe,
>
>>I was going to suggest a differentiating filter but I guess you are "on" to
>>that. Could you post the title/publisher of Frekring's book?
>
>>73's
>>Don
>
>

---

37. Re: [HFSIG:1959] Re: The ARRL Letter Vol 16, No 5

From: Mike - WB0IKT <crucis@sky.net>
Date: Tue, 04 Feb 1997 09:42:22 -0600
To: hfsig@tapr.org
Message-ID: <32F758DE.3289@sky.net>
Rendered body type: text/plain

George Cserenyi wrote:
>
> Charles, how did you do that?
> This is the first time, my pine mailer can't read a message.
> Any suggestion by other pine users?
>
> On Mon, 3 Feb 1997, Charles Hoffman wrote:
>
> Date: Mon, 3 Feb 1997 11:00:37 -0600 (CST)
> From: Charles Hoffman <choffman@pelican.davlin.net>
> Reply-To: hfsig@tapr.org
> To: hfsig@tapr.org
> Subject: [HFSIG:1957] Re: The ARRL Letter Vol 16, No 5
> Parts/attachments:
>   1   OK   46 lines Text (charset: ISO-8859-1)
>   2   ~64 lines Text (charset: ISO-8859-1)
> --
>
>   [Part 1, Text/PLAIN (charset: ISO-8859-1 "Latin 1") 46 lines]
>   [Not Shown. Use the "V" command to view or save this part]
>
>   [Part 2, Text/HTML (charset: ISO-8859-1 "Latin 1") 64 lines]
>   [Cannot display this part. Press "V" then "S" to save in a file]
>
> George

He used enbedded HTML that Netscape news and I think IE news supports.
I read it using netscape, EUDORA however saw the underlying html tags.

--

```
=========================================================================
Mike Watson, WB0IKT
Raymore, Missouri, USA
http://www.sky.net/~crucis

 "Meetings are like fungi---there are more growing in dark, damp
  corners."
=========================================================================
```

38. Re: cw debate

```
From: "Emese Tolnai" <temese@hotmail.com>
Date: Tue, 4 Feb 1997 15:48:18 -0800 (PST)
To: k3ft@erols.com
Message-ID: <199702042348.PAA22944@f9.hotmail.com>
Rendered body type: text/plain
```

```
>Since when is defending or supporting Amateur Radio a religion?
>
>I don't understand your reference point.
>
>thanks!
>
>(by the way.. as a point of reference. are you a licensed Amateur?)
>
>chuck k3ft
>
>k3FT@erols.com

Fist, I hold a licence for 7 years, equivalent to US extra class.
Second, don't play dumb,  the problem wasn't wether
"defending or supporting Amateur Radio"
but a cw vs no cw debate in a list which has nothing to do with it.
Hope this gets thru to you. Got it? Comprende? Go to the cw list.
Only religious zelots toot their horn uninvited on every street corner. Take ur
debate elswhere, we don't want it here.
Nuff said
Anita Bhatia VU2HQX
c/o temese@hotmail.com


------------------------------------------------------------
Get Your *Web-Based* Free Email at http://www.hotmail.com
------------------------------------------------------------
```

39. Re: [HFSIG:1972] Re: cw debate

```
From: Chuck K3FT <k3ft@erols.com>
Date: Tue, 04 Feb 1997 23:08:06 -0500
To: hfsig@tapr.org
Message-ID: <32F807A6.6EDE@erols.com>
Rendered body type: text/plain
```

```
To the sig.. I replied to this gentleman via PRIVATE EMAIL and did not
give him the permission to repost in the public sig. I apologize for him
doing so. Thank you for the patience.

(FYI - I wish he'd have, at least, identified me as the writer of those
words. The rest are his and I do not subscribe to them)

73

chuck K3FT

Emese Tolnai wrote:
>

NO HE DIDN'T I (K3FT) did)
> >Since when is defending or supporting Amateur Radio a religion?
> >
> >I don't understand your reference point.
> >
> >thanks!
> >
> >(by the way.. as a point of reference. are you a licensed Amateur?)
> >
> >chuck k3ft
> >
> >k3FT@erols.com
> (the following are his words,not mine)
```

40. Re: [HFSIG:1973] Re: cw debate

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 4 Feb 1997 23:35:44 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199702050735.XAA06928@servo.qualcomm.com>
Rendered body type: text/plain

I would like to ask the participants on this list to voluntarily refrain from debating the code requirement in this forum. This is ham radio's own answer to the US abortion debate. Like abortion, the CW debate always seems to generate far more heat than light.

Anyone who knows me knows that I have strong opinions on the subject, which I would be happy to express in private email or another appropriate forum to anyone who is interested. But I think we should keep this list focused on its stated purpose, which is the development of advanced digital modulation and coding methods for HF.

I've heard many people express surprise at the consistently high signal-to-noise ratio on this mailing list. That's something I'd like to maintain if at all possible.

Phil

41. Re: cw debate

From: k5cnf1@juno.com (Richard B Morrow)
Date: Wed, 5 Feb 1997 19:25:04 PST
To: hfsig@tapr.org
Message-ID: <19970205.200647.4327.1.k5cnf1@juno.com>
Rendered body type: text/plain

Phil, I agree with you. I have heard debates of ssb vs am, rtty vs cw, as nauseum since I got my call back in 1955. There is enough space to allow everyone to do as they see fit when it comes to modes of communication, so anyone who wants to argue with me please step outside and yell loudly your complaints and I will answer when I hear them.

73, K5CNF

42. Re: [HFSIG:1963] SSTV demod

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 05 Feb 1997 21:15:23 -0800
To: hfsig@tapr.org
Message-ID: <199702060506.VAA28923@PEAK.ORG>
Rendered body type: text/plain

Hi Moe,

>Hi,
>I've been playing around with audio FM detection of SSTV signals using
>DSP. What would be the best method for demodulation? I've tried the
>simple limiter/discriminator which works but seems to underutilize the
>advantages of DSP. Initial attempts of DPLL's have resulted in too
>narrow of a bandwidth and poor dynamic range. I'm now implementing the
>non-limiting scheme using arctan(I/Q) followed by a differentiator and
>correction filter described in Marvin Frerking's book.   Any suggestions
>before I go down some deadend path?
>
>Thanks.
>
>Moe Wheatley AE4JY
>mwheatle@trsystems.com
>
>

Good question - I am interested too in expert's opinions;

The method you describe apparently is a linear detector. The version that uses the limiter is non-linear. Now the question in my mind is which is better? The non-linear approach has some interesting properties, like the "capture effect",
or the behavior or FM in the presence of strong adjacent channel signals. There are some interesting papers from the 60's era where the math for this behavior is explained, and the impression given that it is the way to go (at least that is
what I understood).

Anyone have an opinion; which is better from a systems behavior point of

view, from
a DSP implementation perspective.

Thanks,

Johan, KC7WW

43. Decisions, decisions

From: Charles Brain <chbrain@dircon.co.uk>
Date: Thu, 06 Feb 1997 10:42:25 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970206104225.00669e10@popmail.dircon.co.uk>
Rendered body type: text/plain

Hello,

I am seeking more advice!

I have just finished coding a Viterbi decoder, which appears to work
(albeit with a k=3 code (at the moment)).

I am now looking at changing it from hard decisions to soft decisions. I am
not too sure how to create the transisition probabilites table.

As I will be deeply interleaving the data before transmission I assume
the channel can be considered as DMC ?

The modulation scheme is 39 parallel tones with DQPSK modulation.

Does anyone have any suggestions as to the best way to map the FEC onto the
modem ?

Should I treat each of the tones as a separate DQPSK channel and then use
a decoder that has 4 paths coming into each state (00 01 11 10). This would
cause considerable processing overhead, but might match the modulation better.

Or should I consider the output of the entire modem as a continous bit stream
of 1s and 0s have a single trellis with 2 paths comming into each state (1,0)?

Puncturing. With a punctured code do you treat the 'missing' bits as bits
with low probabilities and use soft decision decoding, or is there another
method (my book doesn't tell me! in fact it doesn't mention puncturing at all).

Low rates, for very low rates what is the best way to transmit the data.
By low rates I mean high redundancy.

Should I simply convolutionally encode the signal, produce multiply copies of
it and interleave it.

Or should I treat the multiple copies as separate copies
of the same thing, run a separate decoder on each one and somehow compare
the metrics comming out of each decoder for each of the paths ? hence producing
a single most likely output ?

Or would it be better to decode the three outputs seperately then majority vote
the outputs or even majority vote the input before decoding ?

Any one got any ideas ?

- Charles G4GUO

44. listproc info

From: dona@tridelta.com (Don Adams)
Date: Thu, 6 Feb 1997 09:52:30 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <199702061452.JAA16743@sparcy.tridelta.com>
Rendered body type: text/plain

I also belong to the SETI league, (THE ULTIMATE DX)
They're looking to startup an e-mail reflector. They would
like to get some info on how to doit.
 Who does the one here at TAPR?

over and out.... Don wa8qzz

45. Re: [HFSIG:1976] Re: SSTV demod (LONG).

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sat, 08 Feb 1997 10:13:43 -0500
To: hfsig@tapr.org
Message-ID: <32FC9827.6874@worldnet.att.net>
Rendered body type: text/plain

Johan Forrer wrote:
>
> The method you describe apparently is a linear detector. The version that
> uses the limiter is non-linear. Now the question in my mind is which is better?
> The non-linear approach has some interesting properties, like the "capture
> effect",
>
> Johan, KC7WW

There is nothing more nonlinear than arctan and no linear detector will
demodulate an FM signal since it is a nonlinear modulation. Here is the
approach I used in DSP2232/SSTV and I am sorry to say that it was never
used by anyone that I know of to print pictures. I even sent them
C code to display all of the fax,sstv modes and they never used it.
I can't even find that C code anymore ;-(.

(1) Software AGC/Limit: This is NOT a hard limiter. It attempts to
keep the incoming signal at a constant envelope until it falls
below a certain threshold and then gives up. That is, I set a
maximum allowable gain. Remember this is DSP, you do not have
to have a causal signal running your gain level and you are
wasting the power of the DSP to "read the future" if you do so.
You should be passing a gain adjusted sample to the follow on
stages based upon a look at both future and past samples. I
prefer the way constant envelope behaves in the discriminators
to the way square wave behaves (square wave of varying frequency
is what you get when you hard limit). Even if you follow this
with yet another bandpass filter to do high frequency component
removal, I do not like what I see in the follow on processing for
this particular signal.
(2) Design an FIR bandpass filter that meets your design criteria.
In the case of SSTV of the old style 1200-2400 Hz or if you want
to receive the AEA stuff, with the digital stuff below black,
make the design wider. Make sure you allow the filter to be
long enough (#taps) that you have less than 1 dB of ripple in the
pass band (if I can afford the ticks for 0.5 dB, I do that).
I typically use Windowed designs (Blackman window) or use the
least squares design procedure. I almost NEVER use Parks McClellan
since I do not care one whit about wasting zero's in my polynomial
making the stop band equiripple. The windowed and least square
fir designs have very sharp roll off in the stop band as you
proceed away from the desired pass band.
(3) Once you have a bandpass filter designed to meet your performance
requirements, make a Hilbert transform filter using EXACTLY the
same design parameters: length of FIR, frequency/magnitude pairs
exactly the same (this may be called "band edges" and "magnitude
responses" in your design program).
(4) Read an incoming sample stuff and run it into the AGC routine and
following that into the "filtering delay line". Run the bandpass
and Hilbert transform filters over the same delay line, making
darn sure that you remove the "advance the tap to the next input
point" in the DSP software for the bandpass routine and then do
the Hilbert transform filter on the SAME delay line but this time
allow the tap to be advanced in anticipation of the next arriving
signal sample.

You have now created an analytic or complex version of the signal. This
allows you to have a fairly wide open "noise bandwidth" on your PLL
loop filter without having the "sum products" give you a ton of ripple
when what you are really after the "difference products". The signal
is b + ih or b+jh depending on whether you are a mathematician or an
engineer ;-).

Suppose your current guess that the frequency is f and your current
guess
of the phase of the incoming signal is p. Then you believe that the
signal you are looking at is (cos(p),sin(p)) or exp(jp) or
cos(p) + i sin(p). They are all the same thing: one sample of a complex
sinusoid. Take the complex conjugate of this, and COMPLEX multiply
this times the analytic signal you made above with the outputs of the
bandpass and hilbert transform (which I will call b and h).

(5) Form the phase detector and 'lock signal' by doing the complex
multiply

of (b,h) and exp(-jp). This is

(bcos(p)+hsin(p)) + j (-bsin(p)+hcos(p))

^ ^
lock signal phase detector = SIN(pherr)

What the DEVIL is this do you say? It is REALLY simple and intuitive. Suppose you have the phase EXACTLY right. Then b=Acos(p) and h=Asin(p) and you get cos^2(p)+sin^2(p) which is exactly A . If you are off then it is smaller than A. A is the amplitude of the incoming signal. If it is not 1 then what you get is the amplitude of the incoming signal rather than one. This is your lock detect signal for the PLL. When it is near enough to one, or bigger than some threshold, you declare that the PLL is locked.

Since (b,h) or b+jh (your incoming signal) is complex it may be represented
as Aexp(j P) where P (capital P) is the real phase of the incoming signal.
If this is the case then (-bsin(p)+hcos(p)) is EXACTLY Asin(P-p), that is,
it is a sinusoidal phase detector. This phase detector output is the error signal for your loop filter; call it pherr. I am going to presume
that you do not know how to calculate Z transforms, and don't want to know.
You need to have determine gains on the loop equations I am about to give
you. If you are ahead of the signal (p>P), then Asin(P-p) is negative and you will retard the phase and lower the frequency. The opposite if you are behind the signal (p<P). Don't worry about (P-p) > Pi, it quickly falls out of this area when the loop gain is high enough in practice. There are other phase detectors, and some might be worth looking at. Go fetch them from the literature if you want to try them out.

Let a,b be numbers.

(6) We will update our estimate of the frequency as follows

F = F + b * pherr

We will update phase with phase error and use the new frequency to get the phase ready for the next sample, all at the same time.

P = P + F + a*pherr. (This is using the new F).

This is the loop equation for the PLL. There are some critical things that we must guarantee. We want this loop to be "critically damped", fortunately, this is easy to guarantee.

Let b = 0.25*a*a and you are guaranteed the loop is critically damped. Now, go play and experiment with a until you have found "one that works".
You will find it is not that critical. There will be a catastrophic event where the loop filter finally has enough gain to track. Vary your incoming signal over a range of amplitudes until the AGC above just begins to stop amplifying further (that is make the amplitude smaller until AGC stops amplifying further). Make a large enough to track this signal.

F, the frequency above is your FM detected signal. I have found that for AFSK signals, such as SSTV where the occupied bandwidth is a large compared to the sample rate (or Nyquist) that "two filter" discriminators don't work as well as this approach. P is your PM detected signal.

Here is some pseudo C code to do the bandpass filter, hilbert transform, bp[] is the array of bandpass filter coefficients, hilb[] is the array of hilbert transform coefficients, delay[] is the delay line (array) of signal samples. In a DSP, you do NOT need to do all of the very wasteful sliding of the delay line, and even here I could do modular arithmetic and make the delay line a circular buffer but that would mask what the algorithm is doing.

```
p = 0;
F = center of range;
while (agc'd_sample_available)  {
        delay[0] = New_signal_sample;
        b = bp[0]  * delay[0];
        h = hilb[0]* delay[0];
        for(j=filtlen-1;j>0;j--) {
           b += bp[j]  * delay[j];
           h += hilb[j]* delay[j]
           delay[j]=delay[j-1];      /*  This is one of the lines that isn't
present in
                                         DSP code since the hardware does circular buffer
                                         modular arithmetic for array pointers in hardware */
        }

        locksig = b*cos()+h*sin(p);
        phase_err = -b*sin(p)+h*cos(p);
        F = F + b*phase_err;
        /* Typically I constrain the range of F */
        if (F is too big) F = max_freq;
        if (F is too small) F = min_freq;
        p = p + F  + a*phase_err;
        if (p > 2Pi) p -= 2Pi;
        if (p < 0) p += 2Pi;
}

Isn't this trivial?  You will find it much easier to work with
and producing nicer results than the typical real signal PLL
whose operating range is usually severely restricted by the
sum products you don't want.  If you filter the sum products
off then your PLL's operating range is restricted because of
the group delay through this lowpass filter.  You don't
alias sidebands around DC, etc.  It is always nicer to think
of signals "God's way": complex.

F is the detected FM signal of interest.  This same section of
pseudo code can be adapted to make FSK, SSTV, multilevel FSK,
etc. demods that are suboptimal for N-FSK but will give you a
quick hack at one that will work.

I hope this isn't too confusing and that I haven't made too many
errors.

--
*************************************************************************
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.         * Amateur Radio, and Golf. Scouts: Advisor
*
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*
*************************************************************************
```

46. Re: [HFSIG:1977] Decisions, decisions

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sat, 08 Feb 1997 10:35:17 -0500
To: hfsig@tapr.org
Cc: dsp@tapr.org
Message-ID: <32FC9D35.4F72@worldnet.att.net>
Rendered body type: text/plain

Charles Brain wrote:

> Hello,
>
>
> - Charles G4GUO
>

Charles:

The metric you should use for a soft decision metric is

-lnP(S|Z)

which is the negative of the natural logarithm of the conditional
probability of your symbol given that the symbol was acutally Z. The
following assumes that you have quantized the signal to some level
and we don't have to integrate densities to this probability. If the
channel is additive white Gaussian noise, and each sample of the
additive
white Gaussian noise is independent of all the other additive noises
this complicated looking thing is then trivial since the density of the
Gaussian kernel collapses to something simple when you apply the
natural logarithm.

-lnP(S|Z) = -ln exp(-d(S-Z)^2/v) where v is noise variance and
d(S-Z) is the Euclidean distance between the received/demodulated
symbol and the putative constellation point Z. That is, you add
d(S-Z)^2 to your metric for a path if you choose Z as the symbol.
naturally, the worst the path metric, the more likely you are to
reject it. The basis for the Viterbi decoding algorithm, or any
dynamic programming algorithm is that of all the paths coming
INTO a new node, one will have the best path metric. There is
nothing you can do, now or in the future, that will make any
of the other paths coming into a node have a better metric than
the one that is best.

The problem on HF is that this is NEVER the right metric because
the noises (receiver, path, interference) are never Gaussian.
This causes you to get too many errors because you give too little
path metric to bad paths. The Viterbi decoding algorithm does a
much better job when you give an erasure (that is high path metric)
to ALL paths than it does when you give to good a metric to a wrong
path. There is a simple "correction" that I apply to the path
metric that seems to really make a big difference. I will write
that up and give a paper on it with my results at some future date.

I wrote this up in a hurry as I have to go to work. If any catches
any major bugs in this, will you please point them out.

Bob
--

****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
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*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

47. Re: [HFSIG:1980] Re: Decisions, decisions

From: Charles Brain <chbrain@dircon.co.uk>
Date: Sun, 09 Feb 1997 17:45:11 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970209174511.006825c0@popmail.dircon.co.uk>
Rendered body type: text/plain

At 08:08 09/02/97 -0600, you wrote:
>Charles Brain wrote:
>>
>> Hello,
>>
>>
>> - Charles G4GUO
>>
>
>Charles:
>
>The metric you should use for a soft decision metric is
>
>>

Thanks for the help Bob, I have looked a Phil KA9Q's code, how he calculates

a metric for BPSK and I am begining to understand that the metric is dependent
on probabilities which are dependent on the modulation and the noise!

I changed my K3 code to a K7 code and it only runs at 250 bits/s ! Far from
Phils 180kbits a second. Still I have made no effort to optimise it just to
understand how the decoder works.

I will re-write it to make it faster when I have time, if I can get 3kbits/s I
will be happy.

-73 Charles

48. Re: [HFSIG:1976] Re: SSTV demod

From: "Frank Knoll" <fknoll@ibm.net>
Date: Sun, 09 Feb 97 21:56:09 +0100
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <199702092137.VAA59796@pop01.ny.us.ibm.net>
Rendered body type: text/plain

On Wed, 5 Feb 1997 23:10:22 -0600 (CST), Johan Forrer wrote:

>
>Good question - I am interested too in expert's opinions;
>
>The method you describe apparently is a linear detector. The version that
>uses the limiter is non-linear. Now the question in my mind is which is better?
>The non-linear approach has some interesting properties, like the "capture
>effect",
>or the behavior or FM in the presence of strong adjacent channel signals. There
>are some interesting papers from the 60's era where the math for this behavior
>is explained, and the impression given that it is the way to go (at least
>that is
>what I understood).
>
>Anyone have an opinion; which is better from a systems behavior point of
>view, from
>a DSP implementation perspective.
>
>Thanks,
>
>Johan, KC7WW
>
>

I use for FAX/SSTV demodulation the arctan demodulator, but
i do not really calc the arctan, because you dont need it.
Here my description:

 First the incomming signal is filtered by a Fir-Filter then
the signal is mixed with SIN/COS down into the baseband.
Afterwards the I/Q Signals is filtered by a lowpass (also a FIR filter)
and the delayed by one sample in the I/Q channel. The delayed
and non delayed signal is "cross multiplied". The
non delayed Q-Signal is mulitplied by the delayed I-Signal and
the delayed Q-Sig. is multiplied by the non delayed I-Sig. .
Then the results are subtracted. If you need a high
accuracy you have to pass the result of the subtraction
through the arctan calculation, but for FAX/SSTV Demod
you do not need it.

The AGC is relativly simple. You have to calc the Signal Power
(of the I +Q Sig) and multiply the I + Q channel with inverse of it.

Without any artan calc. it is an linear demodulator.

The really hard limiting detector (like in the analog world)
is not useable with a DSP because your sample rate is too low
in comparision of the highest modulated signal frequency.

thats all folks...

best regards
Frank Knoll

==
* Frank Knoll (DL2FCH)
* Spessartstr. 12

```
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*  Germany
*  Tel.: +49 6051 969116
*  E-Mail: fknoll@ibm.net   or  dl2fch@amsat.org
============================================================
```

49. Re: [HFSIG:1977] Decisions, decisions

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 10 Feb 1997 16:23:24 -0800 (PST)
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <199702110023.QAA02581@servo.qualcomm.com>
Rendered body type: text/plain

Charles,

As you know, my Viterbi and Fano decoder packages both contain routines to compute metric tables as a function of the signal and noise amplitudes. I work from first principles by integrating the Gaussian probability distributions.

As it turns out, Viterbi decoders are actually quite insensitive to the precise values in their metric tables. This is a significant practical advantage of Viterbi over Fano decoding, which is much more picky about metric tables and signal levels -- so much so that many practical Fano decoders use hard decisions and pay the resulting 2dB performance penalty rather than risk complete decoder failure if the signal gains and/or metric tables are sufficiently inaccurate.

My routines, although instructive, are actually enormous overkill in practice. Your Viterbi decoder will perform almost as well with the simple integer metrics that are used in many modern hardware Viterbi decoders. Since 3-bit soft-decision samples are so common, a typical metric table looks like this:

```
        0 1 2 3 4 5 6 7

1       0 1 2 3 4 5 6 7
0       7 6 5 4 3 2 1 0
```

The top row is the soft decision sample value, given here in "offset" form. A "0" sample means a very strong "0" symbol from the demodulator, while a "7" sample means a very strong "1" symbol. The values 1-3 indicate successively less confident "0" symbols, while the values 4-6 indicate progressively more confident "1" symbol values.

The following two rows give the branch metrics for the "1" and "0" symbol hypotheses respectively. Quite trivial, as you can see. As long as the input signal gain is properly set, the decoder will perform to within a fraction of a dB of theory.

The optimim signal level into a Viterbi decoder is a function only of the incoming noise level, *not* the signal. (This is an argument against receiver AGCs, by the way). If you set the gain too high, the soft decision samples will "clip" at either 0 or 7 and the decoder will degrade to hard-decision performance, which is about 2 dB worse than optimum soft-decision decoding. Similarly, too low a gain will result in the samples only taking on two midrange values (e.g., 3 and 4), again degrading to hard-decision performance. (A *really* low gain setting could cause the samples to all take on the *same* value, which would be a disaster!)

Qualcomm has a free application note on precisely this topic: "Setting Soft-Decision Thresholds for Viterbi Decoder Codec Application Note". It doesn't seem to be on the web, but we have a page where you can order a copy:

<http://www.qualcomm.com/ProdTech/asic/order.html>

I don't have a copy handy here in my office, but I recall from memory that the optimum symbol quantization spacing is something like .45 of a standard deviation of the received noise level, with a leeway of a few dB on either side.

As Bob McGwier quite correctly pointed out, these metric tables and threshold setting rules all assume Gaussian noise. This is generally a valid assumption *only* on the satellite and deep space channels. Many real-world channels have decidedly non-Gaussian noise characteristics. (By the way, one of the practical advantages of direct sequence spread

spectrum is that it makes interference look like Gaussian noise to a
Viterbi decoder.) If you have impulse noise on your channel, you don't
want a strong noise pulse to become a "super confident" 1 or 0 symbol;
you really want it marked instead as an erasure ("don't know").

One way to do this is to use a nonlinear metric table, where symbols
stronger than a certain threshold are mapped to erasures. This might
work for static crashes and intermittent QRM on HF. Another is to
implement a pulse detection/blanking circuit in the receiver and feed
its output to the Viterbi decoder so that erasures are substituted for
whatever (garbage) values come out of the demodulator during the
pulse. This technique would probably work well with the short radar
pulses found on the 70cm band.

More later on your other questions.

Phil

50. Re: [HFSIG:1976] Re: SSTV demod

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 10 Feb 1997 17:05:25 -0800
To: hfsig@tapr.org
Message-ID: <199702110055.QAA28177@PEAK.ORG>
Rendered body type: text/plain

Hi,

>Good question - I am interested too in expert's opinions;
>
>The method you describe apparently is a linear detector. The version that
>uses the limiter is non-linear. Now the question in my mind is which is better?
>The non-linear approach has some interesting properties, like the "capture
>effect",
>or the behavior or FM in the presence of strong adjacent channel signals. There
>are some interesting papers from the 60's era where the math for this behavior
>is explained, and the impression given that it is the way to go (at least
>that is
>what I understood).
>
>Anyone have an opinion; which is better from a systems behavior point of
>view, from
>a DSP implementation perspective.
>
>Thanks,
>
>Johan, KC7WW
>

I seem to have posed my question in an odd way and got a lecture on something
that was interesting, but not quite what I was looking for:

Here it is again:

Are there advantages to using a limiter in an FM demodulator or not ?
Meaning the effects of the limiter's logistic transfer function and its
inherent non-linearity.

Thanks much, 73

Johan

51. Re: [HFSIG:1982] Re: SSTV demod

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 11 Feb 1997 01:58:42 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199702110958.BAA04031@servo.qualcomm.com>
Rendered body type: text/plain

I wonder if anybody has considered a FM demodulator that does an FFT
followed by selection of the strongest frequency bin?

I've not actually tried this idea, but it seems interesting.

Phil

52. Re: [HFSIG:1985] Re: SSTV demod

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Tue, 11 Feb 1997 06:01:20 -0500
To: hfsig@tapr.org
Message-ID: <33005180.3AE1@worldnet.att.net>
Rendered body type: text/plain

Phil Karn wrote:
>
> I wonder if anybody has considered a FM demodulator that does an FFT
> followed by selection of the strongest frequency bin?
>
> I've not actually tried this idea, but it seems interesting.
>
> Phil

Yes I have done this and the problems are maintaining synchrony. For
multitone modems, where you have determining timing, and frequency
offset, it makes for a good approach to doing the filter banks of
complex correlators. For less complex signals, it is a waste. For
FM signals, especially when "noise" or random data is modulating
the signal and the spectrum gets pretty flat, noise+interference can
cause you to pick the wrong bin by miles. The telebit trailblazer
and the Harris D-N-PSK, and multitone FSK modems all use FFT's

Bob

--
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
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*
* (W) 609-279-6240   * Council. Brownie Troop 193
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****

53. Re: [HFSIG:1983] Re: Decisions, decisions

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 11 Feb 1997 14:59:40 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970211145940.0067eba8@popmail.dircon.co.uk>
Rendered body type: text/plain

At 18:25 10/02/97 -0600, you wrote:
>Charles,
>
>
>Phil

Phil thanks for the very helpful reply,

I may tone down my expectations and initially try applying the Viterbi
decoder to an 8ary FSK modem I wrote for the 56002EVM a year or so back.

This would be more of academic interest than of practical use but it is a
start. Whether applying full soft decision decoding to this is necessary,
given the nature of the H.F channel I am begining to doubt.

The modem as well as deciding which of the 8 tones has been received also
produces a pseudo SINAD figure by taking the best tone magnitude and dividing
it by the sum of the magnitudes of all the tones.

I thought that if you know the SINAD that you could work out the BER for the
modulation scheme and hence the probability of receiving the symbol correctly,
this probability could be fed to the decoder to help it on its way.
(Hence my comment about modulation type)

CW interference (by this I don't mean morse code) would of course
really mess this up without some form of excision or whitening filter.

So it may be better to use 1,0 or erasure. Coming to erasures, how do you
tell the decoder it has received an erasure ? Do you simply tell it that it
has received a bit with a 0.5 probability of being right? or is there something
much more subtle than that.

I have read your paper 'Towards new data link protocols' which I found very
interesting, especially how you use the interleaving to puncture the code. My
hope is that a firm protocol can be established so implementations can be
started, by myself or other people.

Extending what I have already discussed onto framing formats.
I have been thinking of using a PN sequence as a sync vector, however I have
recently been thinking of using a number of different PN sequence to indicate
what interleaving/redundancy will follow in the transmission (Gold Codes?).

Then using a ACK/NAK protocol the receiver would in the ACK/NAK also transmit
the best path metric so that the transmitter could decide what FEC to apply
to the next transmission. The metric could be replaced by an increase/decrease
FEC message. I rather like the idea in Clover where you can see what the
receiving end is making of your transmissions. In fact I find staring at that
display far more interesting than the content of most of the QSOs I have !

I also intended using a 32 bit CRC in the data transmissions as
work done by DSTO seems to indicate that CRC16 is not adequate for H.F systems.

I would also like to eventually use a SREJ scheme, I have looked at both
FS1052 DLP and the STC packet protocol for inspiration.

I haven't given up on the parallel tone modem approach, I just haven't
got it to work yet (honesty is the best policy).

As far as SS is concerned I have just about waded through the SS Handbook,
you know the big fat silver coloured one! I have read about full band jammers,
partial band jammers, pulse jammers FEC etc, I still have Vol 2 & 3 to read,
it is a bit like War and Peace! It is also a pretty negative book, the
feeling I came away with was that SS is ideal in certain environments and
under certain hostile conditions but it no panicia and must be applied
properely.

- Thanks Charles

54. Re: [HFSIG:1977] Decisions, decisions

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 11 Feb 1997 11:31:53 -0800
To: hfsig@tapr.org
Message-ID: <199702111922.LAA09548@PEAK.ORG>
Rendered body type: text/plain

Charles,

You may want to look at the 1995 December issue of the Proc
IEE-Communications (the British one not the IEEE) for two papers from the
Darnell group:

a) Coding for both protection and synchronization by Coulton, Hannaford and
Honary.

 and

b)Extended analysis of word synchronization using sequence corellation
techniques
by Grayson and Darnell.

I am sure you are aware of this interesting work.

Trust this may help in your project.

Johan

>
>Hello,
>
>I am seeking more advice!
>
>I have just finished coding a Viterbi decoder, which appears to work
>(albeit with a k=3 code (at the moment)).
>

>I am now looking at changing it from hard decisions to soft decisions. I am
>not too sure how to create the transisition probabilites table.
>
>As I will be deeply interleaving the data before transmission I assume
>the channel can be considered as DMC ?
>
>The modulation scheme is 39 parallel tones with DQPSK modulation.
>
>Does anyone have any suggestions as to the best way to map the FEC onto the
>modem ?
>
>Should I treat each of the tones as a separate DQPSK channel and then use
>a decoder that has 4 paths coming into each state (00 01 11 10). This would
>cause considerable processing overhead, but might match the modulation better.
>
>Or should I consider the output of the entire modem as a continous bit stream
>of 1s and 0s have a single trellis with 2 paths comming into each state (1,0)?
>
>Puncturing. With a punctured code do you treat the 'missing' bits as bits
>with low probabilities and use soft decision decoding, or is there another
>method (my book doesn't tell me! in fact it doesn't mention puncturing at all).
>
>Low rates, for very low rates what is the best way to transmit the data.
>By low rates I mean high redundancy.
>
>Should I simply convolutionally encode the signal, produce multiply copies of
>it and interleave it.
>
>Or should I treat the multiple copies as separate copies
>of the same thing, run a separate decoder on each one and somehow compare
>the metrics comming out of each decoder for each of the paths ? hence producing
>a single most likely output ?
>
>Or would it be better to decode the three outputs seperately then majority vote
>the outputs or even majority vote the input before decoding ?
>
>
>Any one got any ideas ?
>
>- Charles G4GUO
>
>
>

55. Re: [HFSIG:1987] Re: Decisions, decisions

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 11 Feb 1997 12:24:58 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199702112024.MAA19900@servo.qualcomm.com>
Rendered body type: text/plain

First of all, a correction to yesterday's post on soft-decision decoding.

I found my copy of the Qualcomm ap note on setting soft-decision
thresholds. The optimum setting of the A/D thresholds for a Viterbi
decoder with 8 input levels (3-bit quantization) on a symmetric BPSK
channel with Gaussian noise is as follows:

T = n * sqrt(N0/8), n = -3, -2, -1, 0, 1, 2, 3

where N0 is the noise power spectral density.

Re Charles' 8-ary FSK modem,

>The modem as well as deciding which of the 8 tones has been received also
>produces a pseudo SINAD figure by taking the best tone magnitude and dividing
>it by the sum of the magnitudes of all the tones.

A more optimum approach is to compute a separate soft decision value for
each of the three decoded bits as follows.

Assume a mapping of 3 data bit groups to one of 8 tones as follows

Data   tone number
bit
123

000   0
001   1

```
010     2
011     3
100     4
101     5
110     6
111     7

After doing the FFT and obtaining the 8 tone energies, you determine
the soft decision value for bit 1 by summing the energies for bins
4,5,6,7 and subtract the energies in bins 0,1,2,3. To determine the
value for bit 2, add bins 2,3,6,7 and subtract bins 0,1,4,5. To
determine the value for bit 3, add bins 1,3,5,7 and subtract bins
0,2,4,6.

Get the idea? See why it works?

I'll leave the derivation of optimum decision thresholds for this case
to the reader....

Phil
```

56. Re: [HFSIG:1987] Re: Decisions, decisions

```
From: Phil Karn <karn@qualcomm.com>
Date: Tue, 11 Feb 1997 21:51:44 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199702120551.VAA21013@servo.qualcomm.com>
Rendered body type: text/plain

>So it may be better to use 1,0 or erasure. Coming to erasures, how do you
>tell the decoder it has received an erasure ? Do you simply tell it that it
>has received a bit with a 0.5 probability of being right? or is there something
>much more subtle than that.

Yup, you use a sample value that means an equal probability of a "1"
and a "0".

I do note that according to our spec sheets there seems to be no such
3-bit sample value for our hardware Viterbi decoders. But I might be
interpreting them wrong.

In my software decoder there is no problem in establishing a symbol
value that means "erased", since you can specify your own metric
table. Even without a demodulator that can indicate erasures, such a
value is important when implementing punctured codes. (The hardware
decoders can puncture internally and automatically, so you don't need
to enter the erasures yourself.)

Phil
```

57. Re: [HFSIG:1990] Re: Decisions, decisions

```
From: Rolf Sommerhalder <rs@ife.ee.ethz.ch>
Date: Wed, 12 Feb 1997 08:13:37 +0100 (MET)
To: hfsig@tapr.org
Message-ID: <199702120713.IAA02381@gillespie.ee.ethz.ch>
Rendered body type: text/plain

Phil wrote:

> I do note that according to our spec sheets there seems to be no such
> 3-bit sample value for our hardware Viterbi decoders. But I might be
> interpreting them wrong.

There are special codes for feeding erasure symbols to
hardware Viterbi decoders like Qualcomm's Q0256
(see http://www.qualcomm.com/ProdTech/asic/viterbi.html):
If softdecisions are input by hardware signals, then erasure symbols may
be input by specials _pins_ (R0Erasure, R1Erasure, R2Erasure).
If softdecisions are input using writes to Decoder Data Input Registers,
then there are special _bits_ (again called R0Erasure, R1Erasure, R2Erasure)
to indicate erasure symbols. A chip designer in fact called them once
"3.5 bit softdecisions"...
Similar arrangements are provided by chips of other manufacturers too.

73,
Rolf, HB9CWP
```

58. Re: [HFSIG:1991] Re: Decisions, decisions

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 12 Feb 1997 01:24:53 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199702120924.BAA21550@servo.qualcomm.com>
Rendered body type: text/plain

Rolf,

Thanks. I'm blushing because you knew something about my own company's
product that I didn't know... :-)

But then again, I work mostly in software...

Phil

59. [HFSIG:1992] Re: Decisions, decisions -Reply

From: Dave Maples <Dave.Maples@ssc.nasa.gov>
Date: Wed, 12 Feb 1997 06:55:46 -0600
To: hfsig@tapr.org
Message-ID: <s3016ac4.004@wpogate.ssc.nasa.gov>
Rendered body type: text/plain

I have been following this thread for a bit and I must admit it's beyond me.
I graduated way back in '78 from a fairly small school and I am somewhat
(!) behind in this area. My current employment doesn't push me in this
area either, although I can see it coming in the future.

Can you recommend some texts (not necessarily light) in the area of
digital modulation, etc.?

Thanks a lot for your trouble.

60. Re: [DSP:334] Amtor/PActor, etc for EVM

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 13 Feb 1997 09:37:14 -0800
To: dsp@tapr.org
Cc: hfsig@tapr.org
Message-ID: <199702131727.JAA22801@PEAK.ORG>
Rendered body type: text/plain

Hi Doug/Eduardo/Bob,

I feel a dilemma when discussing HF digital topics here and suggest to move
the discussion over to HFSIG - all the experts are really tuned in there.
Perhaps we should let this list remain devoted to DSP in general,
satellite/UHF demodulators etc., and give prospective EVM experimentors a
chance to get started. Does that sound reasonable, or what is the list's
opinion? Tim? Greg?

I will cross-post this reply then if folks dont mind.

Just note (rather long) about what I know of the availability of
demod/decoder software for PC's:

Tom Sailer has a very nice (in my opinion) PC-based software package that
allows RTTY/AMTOR/PACTOR-I. Check his homepage, also the HFSIG software
repository. Some folks have indicated that the user interface is overly
simple, but it works - and it is free and available. Tom's program will work
with a number of demodulators like the ST-6 or DJ6HP filter-type; either
handle data in parallel or serial. For Pactor-I with memory ARQ, the
parallel interface is mandatory. This program also supports a number of DSP
modules like the TI DSK or Juergen Hasch's TI320C25 board.

I wrote PCTOR years ago to work with the AN-93, and it has built-in DSP
support for the TI DSK as well. I have used it successfully with the EVM for
AMTOR. The Pactor side of it, called PC-Pactor, is receive only. The PC
soundcard version of the this software is called "PSATOR" and it does the
full complement of AMTOR/Pactor-I with full transmit and receive. The source
code for PSATOR is available in the HFSIG repository as TORLIB1.ZIP. If
anyone has the inclination to look at it or port it for use with the EVM,
feel free to download it. The DSP modem of PSATOR is an adaptive 100/200
baud modem, the theory and operation was published in QEX - source code
included as well.
BTW, I donated a DSP soundcard to Tom for doing a port of his software to
the soundcard platform. Tom did, and that also works quite nicely - also is
in the HFSIG file repository among other places.

HAMCOM3 is also a good possibility for beginners - it uses a rather simple
zero-crossing detector that works OK on good signals, but I am not sure
whether it can handle a "real modem". Does anyone know perhaps?

It would be relatively easy to write a "remodulator"-type of demodulator on
the EVM that can make HAMCOM work much better.

Regarding the workings of my Pactor decoder: that does what Bob is
describing - a postiori evaluation of every bit over the full packet burst.
It uses a 10-times oversampled data stream which seem to make a difference
in extreme cases of multipath distortion. Actually this is done regardless
of knowing whether the data is 100 or 200 baud, or whether the polarity is
positive or inverted. It does, however has substantial CPU load impact.
Unfortunately the code was written to work with the ADSP-2115 DSP on the
soundcard where the master clock timing originates.
I have toyed with porting it to the EVM, but the speed of the EVM's serial
interface is the problem. One ideally need a higher bandwidth "pipe" between
the DSP and the protocol processor to make this work well.

The latter would not be a problem to obtain permission to release the work,
however, one thing at a time.

Hope this helps a bit - let's continue the thread on HFSIG if anyone needs
to talk about HF DSP theory/implementations. We are really in need of people
that are willing to contribute to developing future HF communications
demodulators and protocols. With that I mean helping out with developing
code besides theorizing about it.

Thanks much for your patience and sorry for wasting bandwidth on this list.

73's

Johan

>
>I agree that it would be really nice to have Amtor and Pactor software
>for the EVM. (RTTY already exists.).
>
>Can you tell me more about "Tom Sailer's HF digital modes terminal
>software for DOS", and the DJ6HP convertor?
>
>Several years ago, Johan Forrer wrote a package called "PCTOR"
>and "PSATOR" that did Amtor and Pactor on the PC, using a simple
>parellel-port-style modem interface. Possibly this program can be
>fixed up a bit to work with the EVM. The problem is that the
>program needs to use seperate interface signals for Tx data, Rx
>data, and the PTT function. Since the EVM has only a two-line
>serial port connection, these signals would have to be
>multiplexed into a proper serial format. Probably it would be
>better if the interface protocol was changed to let the EVM
>instead of the PC handle the bit timing.
>
>Maybe Johan can tell us what he thinks of this idea...
>(I have never actually run his software, just read the documentation.)
>
>Doug
>SV/N1OWU
>
>

61. Re: [HFSIG:1994] Re: [DSP:334] Amtor/PActor, etc for EVM

From: George Cserenyi <zsolt@direct.ca>
Date: Thu, 13 Feb 1997 11:47:45 -0800 (PST)
To: Johan Forrer <forrerj@peak.org>
Cc: hfsig@tapr.org
Message-ID: <Pine.LNX.3.95.970213111930.17119A-100000@cirmos.ci.ca>
Rendered body type: text/plain

Hi,
        was reading Johan's comments on the subject with great interest.
As usual, the answers prompted more questions.
Some time ago there was quite a flurry of articles on this list
regarding pactor-2, what happened, is there someone who's working on it?
I've been operating the SCS pactor-1 PTC for a number of years, that is
NOT dsp based, but consistently outperformed the Kantronics and AEA
products in pactor mode, especially at poor hf condx. Surely we could
match or surpass that with the EVM?
Noticed, some people have more then one EVM boards, are they using
these such a way that #1EVM used as a tuneable filter,

#2EVM as a packet modem, #3EVM as a tnc?
How far can we push a single EVM, can it do all the 3 functions mentioned
above with a reasonable degree of efficiency?
Tongue-in-cheek comment: when I wrote my first hand assembled rtty & cw
program for the Motorola 6800 EVM, (seems like decades ago)
I started with the TX part, too.
Guess the reception and the decoding is still the harder part.
When are we going to see the demod parts of those exotic modems
some people are working on? :)))
My EVM kit can arrive any day now until then,
73,
George
ve7ciz

62. Re: [DSP:334] Amtor/PActor, etc for EVM

From: k5cnf1@juno.com (Richard B Morrow)
Date: Thu, 13 Feb 1997 19:51:24 PST
To: hfsig@tapr.org
Message-ID: <19970213.195336.4535.3.k5cnf1@juno.com>
Rendered body type: text/plain

I am apreciative of what you guys are doing, even though is is out of my
realm of understanding, so keep up the hf stuff, it gives me something
to think about although on a limited amount of ram, my specialty is
antennas. But I intend to start to use some of the newer digital stuff
that you guys figure out and develope

73, K5CNF

63. Re: [HFSIG:1993] Re: Decisions, decisions -Reply

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 14 Feb 1997 10:41:58 +0000
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970214104158.00665770@popmail.dircon.co.uk>
Rendered body type: text/plain

At 07:01 12/02/97 -0600, you wrote:
>I have been following this thread for a bit and I must admit it's beyond me.
>I graduated way back in '78 from a fairly small school and I am somewhat
>(!) behind in this area. My current employment doesn't push me in this
>area either, although I can see it coming in the future.
>
>Can you recommend some texts (not necessarily light) in the area of
>digital modulation, etc.?
>
>Thanks a lot for your trouble.
>
>
>
Hello Dave,

The two books I look at most often are ...

'Communication Systems 3e ' written by Simon Haykin, published by Wiley
ISBN 0-471-30584-7 . This is an undergraduate text book that covers about
everything.

Its chapters include

Introduction.
Representation of Signals and Systems.
Continous Wave Modulation
Random Processes
Noise in CW Modulation Systems
Pulse Modulation.
Baseband Pulse Transmission.
Digital Passband Transmission.
Spread Spectrum Modulation.
Fundamental Limits in Information Theory.
Error Control Coding.
Advanced Communication Systems.

It really covers just about everything and has lots of references. Because
it is a student book it is also relatively CHEAP.

Another book that is worth getting is

'Digital Signal Processing in Communication Systems' by Marvin Frerking
published by Van Nostrand Reihnhold ISBN 0-442-01616-6.

This book describles how to use DSP to design Receivers and Transmitters and modems. It is quite pricey though. The Author works for Rockwell-Collins and it draws on their experiences of H.F systems.

Another fine book is TAPR's own NEW book, see TAPR Homepage for further details www.tapr.org. My copy is in the post at the moment so I can't comment on it yet.

I also have a number of other more specialist books including an Excellant one of FFTs and their applications and some of the Classics. But the above 3 books I
can recommend, for the beginner that has had some formal maths education, if you understand the basics of Calculus but can't do it any more like me you should be able to understand this stuff. It is 15 years ago since I did my degree in Electronics and I have spent my entire working life writing telecoms S/W so
my theory is rusty as well.

Another place to look is the major DSP manufactures WWW pages especially Analogue Devices, they have electronic copies of their applications books on line, although not totally radio related they are useful and *FREE*.

- Hope this helps

Charles G4GUO

64. [HFSIG:1995] Re: [DSP:334] Amtor/PActor, etc for EVM -Reply

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Fri, 14 Feb 1997 13:51:48 +0200
To: hfsig@tapr.org
Message-ID: <s3046c96.065@KIDD.CO.ZA>
Rendered body type: text/plain

I am very interested in any info on PACTOR. I would like to write something for the EVM. I have done a RTTY modem but that is old hat now.

So if anybody has some info please let me know.

Danie zs6awk

>>> George Cserenyi <zsolt@direct.ca> 13/February/1997 08:33pm >>>
Hi,
was reading Johan's comments on the subject with great interest.
As usual, the answers prompted more questions.
Some time ago there was quite a flurry of articles on this list regarding pactor-2, what happened, is there someone who's working on it?
I've been operating the SCS pactor-1 PTC for a number of years, that is NOT dsp based, but consistently outperformed the Kantronics and AEA products in pactor mode, especially at poor hf condx. Surely we could match or surpass that with the EVM? Noticed, some people have more then one EVM boards, are they using these such a way that #1EVM used as a tuneable filter, #2EVM as a packet modem, #3EVM as a tnc?
How far can we push a single EVM, can it do all the 3 functions mentioned above with a reasonable degree of efficiency?
Tongue-in-cheek comment: when I wrote my first hand assembled rtty & cw program for the Motorola 6800 EVM, (seems like decades ago) I started with the TX part, too. Guess the reception and the decoding is still the harder part.
When are we going to see the demod parts of those exotic modems some people are working on? :)))
My EVM kit can arrive any day now until then,
73,
George ve7ciz

65. [HFSIG:1997] Re: Decisions, decisions -Reply -Reply

From: Dave Maples <Dave.Maples@ssc.nasa.gov>
Date: Fri, 14 Feb 1997 07:27:49 -0600
To: hfsig@tapr.org
Message-ID: <s304157e.014@wpogate.ssc.nasa.gov>
Rendered body type: text/plain

Thanks a lot for the book titles. That was just what I was looking for.

66. TAPR comms book

From: Robert.Glassey@nmp.nokia.com
Date: Fri, 14 Feb 97 12:51:25 +0000
To: hfsig@tapr.org

Message-ID: <H0000292035e140d@MHS>
Rendered body type: text/plain

> Another fine book is TAPR's own NEW book, see TAPR Homepage for
> further details www.tapr.org. My copy is in the post at the moment so
> I can't comment on it yet.

I've received mine, and I must say **Congratulations** to Tom for such a
fine book! It is very easy to read, with a straight forward common sense
approach, lots of graphs and analogue equivilents, covers almost
everything, in exactly the right level of detail, even down to how to
write the code and spreadsheets on disk to calculate filter coefficents
etc. There's lots of references if you want to know more, but it covers
everything you need to understand it from a beginners level, with
appedicies covering the maths, right through to actually writing the
code to do it.

Highly recommended!

Rob, OH2/G0VTQ/P

67. Re: [HFSIG:1995] Re: [DSP:334] Amtor/PActor, etc for EVM

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Fri, 14 Feb 1997 16:15:27 +0100
To: hfsig@tapr.org
Message-ID: <3304818F.6517@ife.ee.ethz.ch>
Rendered body type: text/plain

> Some time ago there was quite a flurry of articles on this list
> regarding pactor-2, what happened, is there someone who's working on it?
As far as I know, the Pactor-2 mode isn't fully specified.
There were some marketing hype papers, but I haven't seen
any one providing enough technical information to allow someone
to implement pactor 2. And IMHO reverse engineering it simply isn't
worth doing, as IMHO ptc-2 leaves things to be desired, eg.
robust connect...

73s
Tom

68. Re: [HFSIG:1995] Re: [DSP:334] Amtor/PActor, etc for EVM

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Fri, 14 Feb 1997 16:21:07 +0100
To: hfsig@tapr.org
Message-ID: <330482E3.5F59@ife.ee.ethz.ch>
Rendered body type: text/plain

> Noticed, some people have more then one EVM boards, are they using
> these such a way that #1EVM used as a tuneable filter,
> #2EVM as a packet modem, #3EVM as a tnc?
> How far can we push a single EVM, can it do all the 3 functions mentioned
> above with a reasonable degree of efficiency?

Juergen Hasch DG1SCR is currently working on a 'single evm solution'
for packet radio. His solution uses an EVM for signal demodulation
and bit clock recovery. The HDLC and AX.25 stuff is done by
the host PC. This is a reasonable solution. HDLC and AX.25 can
be done in a modern host CPU with almost no load, and DSPs
like the 56k aren't well suited for the implementation of a
communications protocol. For example they lack byte addressability
which makes you either waste buffer memory or having complicated
and inefficient buffer access.

73s
Tom

69. Re: [HFSIG:2000] TAPR comms book

From: Kory Hamzeh <kory@avatar.com>
Date: Fri, 14 Feb 1997 07:58:58 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <Pine.BSI.3.91.970214075843.26045A-100000@avatar.avatar.com>
Rendered body type: text/plain

What is the name of the book?

On Fri, 14 Feb 1997 Robert.Glassey@nmp.nokia.com wrote:

> > Another fine book is TAPR's own NEW book, see TAPR Homepage for

> > further details www.tapr.org. My copy is in the post at the moment so
> > I can't comment on it yet.
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> fine book! It is very easy to read, with a straight forward common sense
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> everything, in exactly the right level of detail, even down to how to
> write the code and spreadsheets on disk to calculate filter coefficents
> etc. There's lots of references if you want to know more, but it covers
> everything you need to understand it from a beginners level, with
> appedicies covering the maths, right through to actually writing the
> code to do it.
>
> Highly recommended!
>
> Rob, OH2/G0VTQ/P
>
>

70. Re: [HFSIG:2002] Re: [DSP:334] Amtor/PActor, etc for EVM

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 14 Feb 1997 08:56:27 -0800
To: hfsig@tapr.org
Message-ID: <199702141646.IAA21491@PEAK.ORG>
Rendered body type: text/plain

Tom,

>> Noticed, some people have more then one EVM boards, are they using
>> these such a way that #1EVM used as a tuneable filter,
>> #2EVM as a packet modem, #3EVM as a tnc?
>> How far can we push a single EVM, can it do all the 3 functions mentioned
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>
>Juergen Hasch DG1SCR is currently working on a 'single evm solution'
>for packet radio. His solution uses an EVM for signal demodulation
>and bit clock recovery. The HDLC and AX.25 stuff is done by
>the host PC. This is a reasonable solution. HDLC and AX.25 can
>be done in a modern host CPU with almost no load, and DSPs
>like the 56k aren't well suited for the implementation of a
>communications protocol. For example they lack byte addressability
>which makes you either waste buffer memory or having complicated
>and inefficient buffer access.

I know Juergen does excellent work.

Might be a suggestion to look at the Alefnul LEONID that runs on the EVM. It has all the HDLC and KISS code done already and it is the basis for several of the modems that are being used; 1200 FSK, 9600 G3RUH, and the multitone parallel HF modem that Pawel, SP9VRC wrote for us.

However, I am certain Jurgen knows this already, perhaps he is planning on doing something new and innovative? Sounds interesting.

Johan.

71. Re: [HFSIG:1998] Re: [DSP:334] Amtor/PActor, etc for EVM -Reply

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 14 Feb 1997 08:56:28 -0800
To: hfsig@tapr.org
Message-ID: <199702141646.IAA21496@PEAK.ORG>
Rendered body type: text/plain

HI Danie,

>I am very interested in any info on PACTOR. I would like to write
>something for the EVM. I have done a RTTY modem but that is old hat
>now.
>
>So if anybody has some info please let me know.
>
>Danie zs6awk

Look at the SCS web page - there might be some information on the Pactor I protocol.

http://www.scs.com

A little rough on the edges, when you want to write your own especially when it comes to the intricate details of what happens when the link turns around and the speed changes at the same instant from 10ms bits to 5ms bits or vice versa. You can expect lots of fun.

Wasnt it for Tom, HB9JNX, that helped me when I wrote my version some time ago, I probably would still be trying to puzzle it out :-)

73, Johan.

72. DSPX232.ZIP

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sat, 15 Feb 1997 08:17:41 -0500
To: dsp@tapr.org, hfsig@tapr.org
Message-ID: <3305B775.B5C@worldnet.att.net>
Rendered body type: text/plain

I have uploaded DSPx232.ZIP. It contains the bootloader files, and utilities written by Andy Madsen, formally of AEA, to make the ROM's for the DSPX232. This zip does not contain any modems. That will happen
when I return from my trip. There are things I would like to make major improvements in and will do so. The files do contain the programming model for the DSP side of the DSP2232. The modems I know that I can make major improvements in will be primarily the HF modems. I need to replace the guts of them without changing their external functionality. This will allow you to qload these modems to RAM. The RAM in the DSPX232
is battery backed so it will retain these modems 'til the next time you have to do a reset.

The development of these modems will be done on the EVM. I am extremely impressed with the unit. Pat Spatafore tried to sell me on using the Crystal Semiconductors Codec's when we first started. The only reason I did not do it is you could not set different rates in the two channels and that defeated the versatility we wanted for the DSP2232. Since we never UTILIZED the DSP2232's full versatility, . . .

The file is in dsp/upload until someone moves it. I am really sorry that
this is happening two months later than I had hoped it would. Things are
going well at work for me and this has forced me to do dog and ponies and
travel. For that, I am not sorry.

Bob
--
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers, Astronomy*
* 64 Brooktree Rd. * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520 * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963 * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061 * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240 * Council. Brownie Troop 193
*
****

73. [HFSIG:1989] Re: Decisions, decisions

From: Robert.Glassey@nmp.nokia.com
Date: Mon, 17 Feb 97 09:24:02 +0000
To: hfsig@tapr.org
Message-ID: <H00002920360447a@MHS>
Rendered body type: text/plain

> A more optimum approach is to compute a separate soft decision value
> for each of the three decoded bits as follows.
>
> Assume a mapping of 3 data bit groups to one of 8 tones as follows

[snip.. numbering bins 0-7 with binary 000-111]

> After doing the FFT and obtaining the 8 tone energies, you determine
> the soft decision value for bit 1 by summing the energies for bins
> 4,5,6,7 and subtract the energies in bins 0,1,2,3. To determine the
> value for bit 2, add bins 2,3,6,7 and subtract bins 0,1,4,5. To
> determine the value for bit 3, add bins 1,3,5,7 and subtract bins
> 0,2,4,6.
>
> Get the idea? See why it works?
>
> I'll leave the derivation of optimum decision thresholds for this case
> to the reader....

OK, I recall you mentioned this to me once before. I tried it, and sure
it corrected errors compared to data at the same raw rate, but for the
same user rate (comparing to half speed data without FEC) it gave no
improvement in sensitivity, although above the threshold the error rate
dropped steeply. I recall you said this was because of non-coherent
combining losses. At that point I left MFSK since I wanted to improve
sensitivity through FEC.

Is this really the case? It looks to me more like my decoder (K=5) was
working like a hard decision decoder. How does one use this sum of
powers result as a metric? I made a least squared decision to determine
the path through the trellis (it worked on my PSK decoder) looking at
the error between the expected symbol for a given transition and the
received symbol. One book I had that mentioned Viterbi decoding didn't
use probibility functions, but rather described the whole decision
process as a least squared decision. Are the two equivilent?

I've just got this gut feeling that Viterbi decoding should work better
for MFSK than it did in my decoder.

Cheers,

Rob, OH2/G0VTQ

74. RE: [HFSIG:2006] DSPX232.ZIP

From: "Borcher, David (NG-MS)" <DBORCHER@nogmail.gi.com>
Date: Mon, 17 Feb 1997 12:33:00 -0800 (PST)
To: hfsig <hfsig@tapr.org>
Message-ID: <3308B255@gismtpgate.gi.com>
Rendered body type: text/plain

Bob,

(Sorry to use the SIG, don't have your direct Email address)

I'm a newbie to the AEA DSPx controller line.
Is there any commonality between the DSP232 and the
DSP1232/2232 in terms of modem improvement and/or
ROM internals?

Thanks,
Dave K7NG
dborcher@nogmail.gi.com

75. Re: [HFSIG:1872] longfish URLs

From: Frode Weierud <Frode.Weierud@cern.ch>
Date: Tue, 18 Feb 1997 10:29:14 +0100 (MET)
To: hfsig@tapr.org
Message-ID: <Pine.A41.3.95a.970218084741.73882A-100000@sp064>
Rendered body type: text/plain

On Sat, 21 Dec 1996, Charles Brain wrote:

> I am sorry about getting the URL wrong. To prevent anyone wasting their time
> searching this site, the stuff about modems etc (and a mention
> of amateur radio AX.25!) is located in the Longfish Network Handbook area
> under protocols.
>
> URL http://www.itr.unisa.edu.au/~dstowww/longfish/
>

There is a long time since Charles wrote about the longfish URL, but as I
am usually rather slow it is first in these days I visited this site and
then by a completely different route. Sorry Charles, but such is Web
surfing. However, I highly recommend you to visit this site and specially
have a look a Stephen Cook's paper "Advances in High-Speed Modem Design".

The URL for this paper is :
http://charli.levels.unisa.edu.au/~dstowww/longfish/hf95_6.html

The machine charli.levels.unisa.edu.au is the same as www.itr.unisa.edu.au
given by Charles.

This paper was a real eye-opener for me as it gives a very good comparison
between multi-tone and single tone modems and clearly shows that the
multi-tone modem is not yet dead. It is clear that the multi-tone modems as
we know them, based on MIL-STD-188-110A, are outdated with respect to the
coding they are using and that with better coding they can perform as well
if not better than the single tone modem under certain conditions. Stephen
Cook also deals with the questions of how these modems perform when
subjected to impulse noise and co-channel interference. These are
questions I know has been raised on HFSIG earlier.

Concerning the coding question you might like to have a closer look at
Schlegal.C and Costello D.J, "Bandwidth Efficient Coding for Fading
Channels: Code Construction and Performance Analysis", IEEE Journal on
Selected Areas in Communications, Vol.7, 1989, pp. 1356-1368. I have so
far only had a cursory glance at this paper, but it appears to be worth a
much closer study.

As you perhaps know there is a renewed interest in OFDM signals due to the
introduction of terrestrial, digital television. Terrestrial, digital TV
transmissions plan to use a large number of OFDM carriers, 2048 carriers
with 16 QAM is one of the schemes. There is therefore a renewed research
effort going on in this field and I have recently come across a few
referenced which are worth a closer study. They are both in IEEE
Transactions on Consumer Electronics, Vol. 42, No.3, August 1996.
Qun Shi, "OFDM in Bandwidth Nonlinearity" pp.253 - 258, deals with the
in-band bit-error-rate degradation and induced adjacent channel
interference as a result of amplitude limiting or clipping. The other
paper is Claus Muschallik, "Improving an OFDM Reception Using an Adaptive
Nyquist Windowing", pp.259 - 269. This paper is the very interesting as it
show the effect of using windowing on the FFT filter bank.

I have also had a closer look at the reports (STANAGs) on Peter Reynold's
Home page that Hakan Bergzen mentioned earlier. The URL for his page is
http://ourworld.compuserve.com/homepages/pdreynolds/

The Technical Note STC TM-940, which is very long 174 pages, and which
mainly deals with the subnet architecture for data transmission over HF
radio has some interesting comparisons of the MIL-STD-188-110A serial tone
modem and the serial tone modem in STANAG 4285. To my surprise they show
that the MIL-STD-188-110A serial tone modem is quite a bit better than
STANAG 4285. This comparison is in the subsection "Modem sublayer" which
appears on page 130 -139.

73's Frode, LA2RL

Frode Weierud Phone : +41 22 7674794
CERN, SL, CH-1211 Geneva 23, Fax : +41 22 7679185
Switzerland E-mail : Frode.Weierud@cern.ch

76. Re: [HFSIG:2004] Re: [DSP:334] Amtor/PActor, etc for EVM

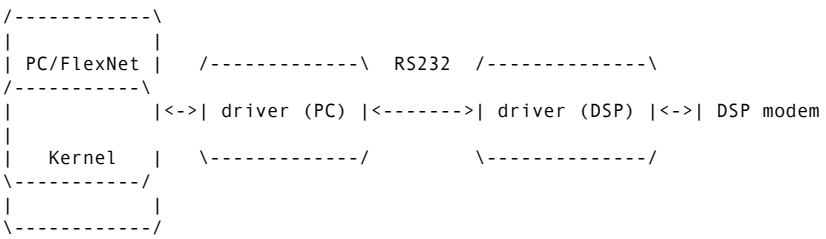
From: Hasch@t-online.de (J□rgen Hasch)
Date: Mon, 17 Feb 1997 20:59:34 -0800
To: hfsig@tapr.org
Message-ID: <33093736.3307@t-online.de>
Rendered body type: text/plain

Hello Johan,

> Might be a suggestion to look at the Alefnul LEONID that runs on the EVM. It
> has all the HDLC and KISS code done already and it is the basis for several
> of the modems that are being used; 1200 FSK, 9600 G3RUH, and the multitone
> parallel HF modem that Pawel, SP9VRC wrote for us.
>
> However, I am certain Jurgen knows this already, perhaps he is planning on
> doing something new and innovative? Sounds interesting.
>

well, I took a good look at leonid and didn't try to invent the wheel
again.

Here's a small overview of what I did an how the driver is working:



The channel access and all timing (except txdelay, which is done in the dsp part) is
determined by the flexnet kernel. A flexnet driver delivers/receives L2 frames to/from
the kernel.
The PC part of the driver only receives the completely decoded frames and puts it in the
right format for the kernel, all the hard work is done in the dsp part of the driver.
If Leonid hadn't already implemented what I needed, I would have put the HDLC stuff in the PC part of driver, as Tom suggested.
The DSP modems are basically the same as with leonid, I only made them linkable and removed
the kiss stuff.

Some further notes:
 - no more DSP 4 card compatibility, mostly because I use the 56002 timer
   for programmable delays
 - no kiss anymore
 - separate source modules, enables linking with DSPLNK. This creates one single .CLD file.
   There is a little utility I've written for downloading CLD files to the EVM,
   called DLCLD, works great!
 - more sampling rates (4,8kHz, 19,2kHz, 38.4kHz) by using a second crystal (14.7456MHz).
 - 4 LEDs on the PB-ports are used (PTT,DCD, connect and status)
   No more heartbeat, I trust that the DSP won't crash that easy. If it crashes,
   I can reset the EVM board with the DTR signal at the RS232 port.

73's de Juergen,
http://ourworld.compuserve.com/homepages/dg1scr/homee.htm

77. MFSK

From: Matt Ettus <mne@cmu.edu>
Date: Wed, 19 Feb 1997 21:21:56 -0500
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970220022156.006a92b8@gauss.ece.cmu.edu>
Rendered body type: text/plain

Hi everyone. I'm new to the list, and I'm sure there's been a lot of conversation about this, but...

        What is the best way to demodulate MFSK on a DSP? Are FFTs best? What about doing the same thing chips like the Motorola 13055 do it (Mixing with a phase shifted version of the same signal) ?

        Also, is there anyone out there who has channel simulator code of any type working on a TI DSK? I know there is some code for the DSP-93, but I don't have access to one, or the source code to the simulator to do the porting.

        Thanks
        Matt Ettus
        N2MJI
        mne@cmu.edu

78. [HFSIG:2006] DSPX232.ZIP -Reply

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Fri, 21 Feb 1997 08:14:33 +0200
To: hfsig@tapr.org
Message-ID: <s30d5fd7.012@KIDD.CO.ZA>
Rendered body type: text/plain

Bob will your zip files contain source as well ? The EVM (56002) is my
primary modem,use it for anything from HF up to satellites :-) Welcome
aboard the EVM fraternity !

Danie

>>> Robert McGwier <rwmcgwier@worldnet.att.net> 15/February/1997
03:26pm >>>
I have uploaded DSPx232.ZIP. It contains the bootloader files, and utilities
written by Andy Madsen, formally of AEA, to make the ROM's for the
DSPX232. This zip does not contain any modems. That will happen
when I return from my trip. There are things I would like to make major
improvements in and will do so. The files do contain the programming
model for the DSP side of the DSP2232. The modems I know that I can
make major improvements in will be primarily the HF modems. I need to
replace the guts of them without changing their external functionality.
This will allow you to qload these modems to RAM. The RAM in the
DSPX232 is battery backed so it will retain these modems 'til the next time
you have to do a reset.

The development of these modems will be done on the EVM. I am
extremely impressed with the unit. Pat Spatafore tried to sell me on using
the
Crystal Semiconductors Codec's when we first started. The only reason
I did not do it is you could not set different rates in the two channels and
that defeated the versatility we wanted for the DSP2232. Since we
never UTILIZED the DSP2232's full versatility, . . .

The file is in dsp/upload until someone moves it. I am really sorry that this
is happening two months later than I had hoped it would. Things are
going well at work for me and this has forced me to do dog and ponies
and travel. For that, I am not sorry.

Bob
-- **
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor   *
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
 *
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a   *
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington   *
* (W) 609-279-6240   * Council. Brownie Troop 193   *
****

79. PC DSP Software Group Purchase

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Fri, 21 Feb 1997 13:59:13 -0600
To: " TAPR/AMSAT DSP "<dsp@tapr.org>, " DSP-93 Build "<dsp-93@tapr.org>,
        "HF SIG list   mailing"<hfsig@tapr.org>,
        "TAPR-BB list   mailing"<tapr-bb@tapr.org>
Message-ID: <v03007808af33aeafafe6@[208.134.134.40]>
Rendered body type: text/plain

This is to let those who were in the PC DSP Software Group purchase know
that the software has arrived and is being shipped.

Cheers - Greg

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd
-----

80. Coil Form

From: mbranda@cftnet.com (Mike Branda)
Date: Sun, 23 Feb 1997 11:29:21 -0400
To: hfsig@tapr.org
Message-ID: <199702231630.LAA13155@renoir.cftnet.com>
Rendered body type: text/plain

I need to find a coil form for an hf trap. The form needs to have an od of
2.5 inches and the ability to hold 9 turns of #14 wire at 6 turns per inch.
Any ideas?

Thanks.

Mike Branda - K4HN

****
* Check out The Homepage at http://www.geocities.com/SiliconValley/5897/ *
****

81. Re: [HFSIG:2013] PC DSP Software Group Purchase

From: Bill Carver <bcarver@magiclink.com>
Date: Sun, 23 Feb 1997 13:23:59 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <199702231923.NAA11202@tapr.org>
Rendered body type: text/plain

I missed the original Motorola EVM board sale. If there is one that didn't get get used I'd be happy to buy it.

Bill - W7AAZ

82. Marine Pactor-II

From: MMRCUSVC@aol.com
Date: Mon, 24 Feb 1997 13:26:17 -0500 (EST)
To: hfsig@tapr.org, RChasteen@maf.org, jerome@dove.com.ph, DPedersen@maf.org
Message-ID: <970224132617_1879831659@emout03.mail.aol.com>
Rendered body type: text/plain

February 24, 1997

To Whom It May Concern:

Mobile Marine Radio continues the development of permissive narrow and wideband data modes, which we hope to introduce as regular service to the maritime radio environment in the near future.

At this time, emissions are FEC, ASCII, and set up for +-85 hertz shift for FSK
from the center frequency, and tones located +-85 hertz reference the center frequency for binary phase shift keying (BPSK) modes. The center frequencies are 4256.0, 6416.0, 8473.5, 13051.5, and 16997.6 kHz.

1. WLO Radio ran test on BPSK short versus long. We timed the length of time
it took to send our "Packet Radio Data Broadcast" information.

Mode	Time To Send (Minutes)
FSK100	4.11
FSK200	1.60
PSK100 SHORT	4.80
PSK100 LONG	2.50
PSK200 SHORT	1.68
PSK200 LONG	1.13
PSK400 SHORT	0.75
PSK400 LONG	0.56
PSK800 SHORT	0.41
PSK800 LONG	0.33

>From the above it appears the long mode moves more data in a shorter time than the short mode.

2. In off the air monitoring of BPSK:

a. With the type of propagation we have between our monitoring site and transmit site the short mode will produce less errors and more printed data when
propagation is poor than the long mode.
b. During the day we are able to copy the 400 short about 99 percent of the time, 400 long about 95 percent of the time, 800 short about 5-10 percent of the
time, and 800 long about 0-1 percent of the time.

3. The PSK time slot broadcast is set up as follows:

a. Traffic list and weather schedule moved up from 35 minutes past the hour (HH+35) to on the hour (HH)

b. Our information runs 400 long from the end of the traffic list/weather to HH+26.

c. From HH=26 to HH+58 the information runs a schedule of different modes

and power levels:
HH+

26	FSK200	
28	100 LONG	
30	200 SHORT	
32	200 LONG	
34	400 SHORT	
36	400 LONG	
38	800 SHORT	
40	800 LONG	
42	400 SHORT	-3DB
44	800 SHORT	-3DB
46	400 SHORT	-6DB
48	800 SHORT	-6DB
50	400 SHORT	-9DB
52	800 SHORT	-9DB
54	400 SHORT	-12DB
56	800 SHORT	-12DB

Note: The above power level changes could tell us (1) whether the transmitter is
overdriven, and (2) how much signal there is.

d. At HH+58 FSK200 serves as a tuning signal for the traffic list and weather
which begins at HH+00. The CW identification switches to FSK100 before sending the station identification, "WLO".

4. WLO Radio welcomes any feedback from all stations copying the broadcast.
Contact me at (334) 660-5110 ext 14 or Tim Urban (Chief Engineer) ext. 4.

Thank you,

Curt Fauver
Assistant to the President
Internet: MMRCUSVC@aol.com
          or WloEmail@aol.com

83. Unsubscribe

From: "Gene & Judy Vajgrt" <gvajgrt@mobis.com>
Date: Wed, 26 Feb 1997 17:22:23 -0600
To: <hfsig@tapr.org>
Message-ID: <199702262324.RAA22594@main.mobis.com>
Rendered body type: text/plain

Please "unsubscibe" me from the HF SIG messages.
Thanks.

84. Re: [HFSIG:2007] Re: Decisions, decisions

From: Phil Karn <karn@unix.ka9q.ampr.org>
Date: Thu, 27 Feb 1997 01:29:02 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199702270929.BAA09524@unix.ka9q.ampr.org>
Rendered body type: text/plain

Rob,

Re soft decision Viterbi decoding for MFSK --

The math here is really, really hairy. Proakis and Wilson show some of the derivations. The energy in each FFT frequency bin containing only noise has a Rayleigh distribution; in the correct bin it has a noncentral chi-square distribution. The detection process involves comparing the sums of these random variables.

The bottom line is that there is an optimum code rate that depends on the number of frequency bins and also on the fading characteristics of the channel. On nonfading channels there is a broad optimum at about code rate 1/2. On Rayleigh fading channels the optimum is much lower, particularly for higher order MFSK alphabets; it can be as low as 1/10.

Your best approach may be to run some simulations, with and without Rayleigh fading, to see what effect code rate and sample scaling have on performance.

Re the Viterbi decoder, yes, it finds the maximum likelihood sequence, which is as far as I know the same as minimizing the least square error.

That is, it tries to maximize the product of the likelihood
correlations between each received symbol and the local regenerated
symbols. In practice it first computes the logarithm of the likelihood
values so it can use repeated addition rather than multiplication. But
the end result is the same.

Phil

85. Re: [HFSIG:2018] Re: Decisions, decisions

From: Glen Worstell <Glen_Worstell@notes.seagate.com>
Date: 27 Feb 97 12:06:13
To: hfsig <hfsig@tapr.org>
Message-ID: <9702271845.AA1633@notes.seagate.com>
Rendered body type: text/plain

>Re the Viterbi decoder, yes, it finds the maximum likelihood sequence,
>which is as far as I know the same as minimizing the least square error.

Can't resist sticking my nose in:

The Viterbi detector finds the maximum likelihood sequence only if
the samples are corrupted by noise which is Gaussian and uncorrelated.
There are also some issues with correlation in the data.

In disk drives (my work) the noise is not Gaussian, and this causes a
very real degradation in the performance of the Viterbi detector compared
to a maximum likelihood sequence detector.

cheers,
Glen, KG0T

86. Unsubscribe

From: Miguel Angel Garcia Estrello <migarcia@csiam1.mxl.uabc.mx>
Date: Thu, 27 Feb 1997 16:25:29 -0800
To: hfsig@tapr.org
Message-ID: <331625F9.7A3D@csiam1.mxl.uabc.mx>
Rendered body type: text/plain

Please "unsubscibe" me from the HF SIG messages.
 Thanks.

87. Re: [HFSIG:2019] Re: Decisions, decisions

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 27 Feb 1997 17:26:19 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199702280126.RAA14274@servo.qualcomm.com>
Rendered body type: text/plain

>Can't resist sticking my nose in:

>The Viterbi detector finds the maximum likelihood sequence only if
>the samples are corrupted by noise which is Gaussian and uncorrelated.
>There are also some issues with correlation in the data.

Quite correct. But if you have correlated noise, you can take
advantage of it in the channel model used by the Viterbi decoder.
This is sometimes done with intersymbol interference, which by definition
is correlated from one symbol to the next.

>In disk drives (my work) the noise is not Gaussian, and this causes a
>very real degradation in the performance of the Viterbi detector compared
>to a maximum likelihood sequence detector.

Again quite correct. Many disk drives use "partial response"
signalling, which is a deliberate, controlled form of intersymbol
interference. If the Viterbi decoder can take it into account, it can
actually do better than on a channel without intersymbol interference.

Even the convolutional encoder for which the Viterbi decoder was
originally designed is nothing more than a device that generates
deliberate intersymbol interference, the extent of which is the
constraint length.

If you're talking about true (non-data-related) noise or interference,
then you're right again that this causes a Viterbi decoder to perform
poorly. Because you generally can't model or predict this type of
degradation, the usual solution is to interleave the channel symbols

to break up correlations between adjacent coded symbols.

Phil

88. Re: [HFSIG:2018] Re: Decisions, decisions

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 27 Feb 1997 17:40:17 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199702280140.RAA14348@servo.qualcomm.com>
Rendered body type: text/plain

A few more remarks on the note I sent (late) last night.

I misspoke a bit on the distributions of the energies in a M-ary FSK receiver bin. Assuming a stationary nonfading gaussian channel, the signal *amplitude* in the correct frequency bin is Ricean-distributed, while the incorrect bins are independent and Rayleigh distributed. The signal *powers* are therefore non-central chi-squared for the correct bin and central chi-squared for the incorrect bins.

The probability of raw symbol error for an uncoded channel using simple max-energy detection is therefore the probability that one or more of the energies in the incorrect bins exceeds the energy in the correct bin. The nonlinearities in the decision process are what makes the math so hairy.

The soft-decision demodulation approach I described earlier is probably a little different. Because you're adding up several uncorrelated random variables to get the decision variable for each channel bit, the central limit theorem kicks in and the distribution of the resulting decision variables approximates a Gaussian distribution.

This means you can still compute the branch metrics for a Viterbi decoder assuming that the noise in the decision variables is gaussian and not be far off. Now determining the actual mean and variance of these decision variables takes some more math, which I don't have in front of me at the moment. Or you can just experiment and find the right values empirically. The Viterbi decoder seems remarkably robust to non-optimum branch metrics anyway.

Phil

89. Unsubscribe

From: "Guillaume Klaaysen" <guillaum@worldonline.nl>
Date: Fri, 28 Feb 1997 22:18:04 +0100
To: <HFsig@tapr.org>
Message-ID: <199702282118.WAA02026@worldonline.nl>
Rendered body type: text/plain

Please "unsubscibe" me from the HF SIG messages.
Thanks.

90. unsubscribe

From: Miguel Angel Garcia Estrello <migarcia@csiam1.mxl.uabc.mx>
Date: Fri, 28 Feb 1997 16:15:30 -0800
To: hfsig@tapr.org
Message-ID: <33177522.7B26@csiam1.mxl.uabc.mx>
Rendered body type: text/plain

Please "unsubscribe" me from the HF SIG messages.
Thanks.

March 1997

Original source file: hfsig.9703
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9703

1. Re: [HFSIG:2023] Unsubscribe

From: Kleber Almeida <kleber@magiclink.com.br>
Date: Sun, 02 Mar 1997 03:16:34 -0300
To: hfsig@tapr.org
Message-ID: <33191B42.6215@magiclink.com.br>
Rendered body type: text/plain

Guillaume Klaaysen wrote:
>
> Please "unsubscibe" me from the HF SIG messages.
> Thanks.

The correct form is:

unsubscribe hfsig@tapr.org.

2. (no subject)

From: "Alberto di Bene (xx39-2-596.25744)" <dibene@VNET.IBM.COM>
Date: Tue, 4 Mar 97 10:43:45 DST
To: hfsig@tapr.org
Message-ID: <199703040946.DAA09292@tapr.org>
Rendered body type: text/plain

Subject:Tunable FIR filters

> Hi fellow dsp progammers...
>
> I have been intrigued* by the DSP CW filters on the latest DSP HF radio's.
> These filters can be varied contineously from say 50Hz to 2000Hz. Can
> anybody give me a clue on how this is done ?
>
> I have written some CW filters for HF for the DSP56002EVM (see cwawk.zip on
> upload) but was using a set of FIR filter coef's for a set of filters. Is
> there perhaps a DSP algorithm let allows one to make a virtually
> contineously variable FIR filter ? I would like to implemented this on my
> 56002 + hf radio and elevate it to 'dsp rig' status :-)) The 56002EVM has
> limited memory so one cannot load many many coef sets.
>
> Yours sincerely
>
> Danie zs6awk

Danie, I have programmed a Texas 'C50 DSK with such an algorithm. It receives from a PC (at present time, in the future it will be an 87C752 microcontroller) the low-cut and the high-cut frequencies, and, while the foreground, interrupt-driven, task computes the output based on the current set of h(n) values, the background task computes a new set of h(n) on-the-fly, based on the input from the PC. I just take the two frequencies, smooth the transition between 0 and 1 with two intermediate values each (to minimize the Gibbs ripple), then compute an inverse FFT on the resulting smoothed frequency-domain window, and, voila, the new set of h(n) is ready. The critical step is to find the two intermediate values between 0 and 1, and between 1 and 0. Some hints can be find in the literature, but I preferred to do a simulation using Matlab, which showed me that those values are somewhat depending on the band limits, so now I use two different sets, according to which low-cut and high-cut frequencies the DSP receives from the PC.

The 'C50 has plenty enough horse power to do all of this (plus an LMS filter, a notch filter and AGC) at 9.6 kHz sampling rate. Usually the commercial DSP boxes use an Analog Devices AD2105, moreover programmed in C instead of optimized assembler, so I doubt they can use a similar method.

73s

Alberto di Bene, I2PHD dibene@vnet.ibm.com i2phd@amsat.org

SYS0753 WIN.COM not found
A)bort, R)etry, C)ongratulate
---

3. [HFSIG:2026] Tunable FIR filters

From: Robert.Glassey@nmp.nokia.com
Date: Tue, 4 Mar 97 15:21:16 +0000
To: hfsig@tapr.org
Message-ID: <H0000292037e9de2@MHS>
Rendered body type: text/plain

> Hi fellow dsp progammers...
>
> I have been intrigued* by the DSP CW filters on the latest DSP HF
> radio's. These filters can be varied contineously from say 50Hz to
> 2000Hz. Can anybody give me a clue on how this is done ?
....
> Danie zs6awk

Hi Danie,

For CW, if the bandwidth is constant and only the centre frequency changes the easiest approach I can think of is what I use in my RTTY modem to move the receiver tone filters anywhere in the audio spectrum.

Here how:

Mix (multiply) the real input samples with cos(2.pi.fc) - j sin(2.pi.fc) then apply a fixed low pass filter of half the bandwidth of the CW filter, then multiply the result again by cos(..) + j sin(..) and take the real component. Since the imaginary component is zero in the first multiply and the resulting imaginary component is ignored in the second multiply, and since the filter coefficents are also real, you can simplify the calculations. The sin and cos waves can be generated from a look-up table, of say 2048 samples using a fractional step value to find the next sin/cos sample. The step value is proportional to freqency and can be defined to any resolution you like (I use 16 bit resolution then divide by 32 to get the table index). You need so many table entries to reduce time quantisation jitter.

This shifts the spectrum down an arbitary amount (preserving resulting negative frequencies since the result is complex) applies the filter around zero-IF then shifts the spectrum back to where it was. (I dont shift it back in my RTTY demodulator.)

Variations on this method can be used to do a high-cut/low-cut variable bandwidth filter by shifting a wide low pass filter around in the audio spectrum to cut the high then low sides, with no need to recalculate filter coefficents (so it's virtually instant).

For CW you could use decimation to reduce the sample rate the FIR filter has to run at (output rate) since the bandwidth is greatly reduced, but then you have to do some interpolation (oversampling) to recreate the removed samples without generating strong sidebands in the filter output. A simple one or two pole IIR low pass filter would be suitible after a linear interpolator to reduce the sidebands. Sidebands are caused by aliasing at the lower sample rate. A 20:1 decimation on a 8kHz sample stream would give 400Hz between sidebands and allow a simple filter to reduce them to a suitable level after oversampling (still at zero-IF) before you shift the signal back to the original audio freqeuncy. The FIR still effectivly runs at full rate, except you ignore (dont calculate) 19 out of 20 output samples, so the input aliasing is filtered by the FIR. This way you can use much longer FIR filters, and get much sharper cutoffs and get better out of band rejection.

Cheers,

Rob

4. HFSIG List

From: Claude Frantz <claude@bauv106.bauv.unibw-muenchen.de>
Date: Mon, 17 Mar 1997 07:10:07 +0100 (MET)
To: hfsig@tapr.org@netwman.RZ.UniBw-Muenchen.de
Message-ID: <199703170610.HAA06354@bauv106.bauv.unibw-muenchen.de>
Rendered body type: text/plain

I'm not more receiving the list since the begin of the month.
What has happening ?

73,
claude

5. Re: [HFSIG:2028] HFSIG List

From: Ulf.Kumm@t-online.de (Ulf Kumm)
Date: Mon, 17 Mar 97 14:54 MET
To: hfsig@tapr.org
Message-ID: <m0w6crj-0007sRC@fwd00.btx.dtag.de>
Rendered body type: text/plain

At 00:24 17.03.97 -0600, you wrote:
>I'm not more receiving the list since the begin of the month.
>What has happening ?
>
>73,
>claude
>
same here, but this msg arrived

73! Ulf, DK9SJ>

Ulf Kumm, DK9SJ,   D-70597 Stuttgart, Johannes-Kraemer-Str. 34
eMail: Ulf.Kumm@t-online.de   Phone: +49 711 7678-925 Fax: -924
Internet homepage: http://home.t-online.de/home/Ulf.Kumm/main.htm
----

6. Re: [HFSIG:2028] HFSIG List

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 17 Mar 1997 07:52:47 -0800
To: hfsig@tapr.org
Message-ID: <199703171552.HAA01325@PEAK.ORG>
Rendered body type: text/plain

Claude,

>I'm not more receiving the list since the begin of the month.
>What has happening ?
>
>73,
>claude
>
>

Not to worry - the list is still working.
It is only a bit quiet at the moment but traffic is bound to pick up again.

73, Johan KC7WW

7. HF antenna for apartment

From: travis@campbellsci.com (Gary Travis Roberts)
Date: Mon, 17 Mar 1997 09:32:49 -07:0
To: hfsig@tapr.org
Message-ID: <19970317163440070.AAA249@travis_roberts>
Rendered body type: text/plain

I currently live in a bottom floor apartment and was wondering what
would be a good antenna for my situation. I don't know if this is
the right place to ask, but what the heck. Thanks in advanced.

Gary

G. Travis Roberts   KC7HHK
MIS Hardware Specialist / Assistant Network Admin
Campbell Scientific, Inc.
815 W 1800 N
Logan, Ut 84321
Phone: (801) 750-9583
FAX: (801) 750-9579
E-mail: travis@campbellsci.com
http://www.campbellsci.com

8. Re: [HFSIG:2031] HF antenna for apartment

From: nt1g@bitwise.net (Skip Flem)
Date: Mon, 17 Mar 1997 14:11:34 GMT
To: hfsig@tapr.org
Message-ID: <17199734.Amiga@mailbox.bitwise.net>
Rendered body type: text/plain

> On Mon, 17 Mar 1997 10:47:44 -0600 (CST) you wrote:

> I currently live in a bottom floor apartment and was wondering what
> would be a good antenna for my situation. I don't know if this is
> the right place to ask, but what the heck. Thanks in advanced.
>
> Gary
>
>
>
> G. Travis Roberts   KC7HHK
> MIS Hardware Specialist / Assistant Network Admin
> Campbell Scientific, Inc.
> 815 W 1800 N
> Logan, Ut 84321
> Phone: (801) 750-9583
> FAX: (801) 750-9579
> E-mail: travis@campbellsci.com
> http://www.campbellsci.com
>
>
You didn't mention your building's method of construction but let's not

talk ourselves out of an idea. 8^)

I live in a wood frame house and this works for me:

A long wire from a SGC antenna tuner up the side of the house and around the perimeter.

The loop doesn't close but stays open for the final half of the front. The gutters are

plastic...

  This could work for a brick building but I thought they used some kind of steel

mesh near the roof line.

 What the heck.. at least you got a response...

de Skip Flem, NT1G <nt1g@bitwise.net> Hull, Massachusetts 02045-0815 U.S.A.
                         N42o18.305' W070o53.144' 5 M ASL (@low tide)

9. unsubscribe

From: sam taylor <simplesimon@InfoAve.Net>
Date: Mon, 17 Mar 1997 18:19:32 -0800
To: hfsig@tapr.org
Message-ID: <332DFBB4.29D9@InfoAve.Net>
Rendered body type: text/plain

I have a bigger problem! I unsubscribed and now I'm getting them again!
Can anyone send that list of commands to unsubscribe again? I won't
delete it for at least a year...thanks

10. Re: [HFSIG:2018] Re: Decisions, decisions

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Wed, 19 Mar 1997 04:25:43 -0500
To: hfsig@tapr.org
Message-ID: <332FB117.3453@worldnet.att.net>
Rendered body type: text/plain

Phil Karn wrote:
>
> Rob,
>
> Re soft decision Viterbi decoding for MFSK --
>
>
> Re the Viterbi decoder, yes, it finds the maximum likelihood sequence,
> which is as far as I know the same as minimizing the least square error.
>
> That is, it tries to maximize the product of the likelihood
> correlations between each received symbol and the local regenerated
> symbols. In practice it first computes the logarithm of the likelihood
> values so it can use repeated addition rather than multiplication. But
> the end result is the same.
>
> Phil

This is NOT correct. It computes the maximum likelihood path as you

said
and it is NOT the same path as minimizing the least square error. The
minimum bit error rate is achieved by computing the symbol at each time
that gives the maximum likelihood given ALL of the observations, i.e.
you
do a posteriori estimation. In a linear channel with Gaussian noise,
THIS
minimizes least square error. It is terribly confusing until you pour
through all of the math associated with it.

Bob
--
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

11. Re: [HFSIG:2019] Re: Decisions, decisions

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Wed, 19 Mar 1997 04:27:37 -0500
To: hfsig@tapr.org
Message-ID: <332FB189.508@worldnet.att.net>
Rendered body type: text/plain

Glen Worstell wrote:
>
> >Re the Viterbi decoder, yes, it finds the maximum likelihood sequence,
> >which is as far as I know the same as minimizing the least square error.
>
> Can't resist sticking my nose in:
>
> The Viterbi detector finds the maximum likelihood sequence only if
> the samples are corrupted by noise which is Gaussian and uncorrelated.
> There are also some issues with correlation in the data.
>
> In disk drives (my work) the noise is not Gaussian, and this causes a
> very real degradation in the performance of the Viterbi detector compared
> to a maximum likelihood sequence detector.
>
> cheers,
> Glen, KG0T

This is the reason that on HF especially, I do a quick and dirty
reestimation
of the noise statistics and use this to modify the metric when I do
viterbi
decoding in HF modems. The performance improvement can be pretty
dramatic
on Rayleigh fading channels and other decidely non-Gaussian channels.

--
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

12. Re: [HFSIG:1963] SSTV demod

From: "Wheatley, Maurice" <mwheatle@trsystems.com>
Date: Thu, 20 Mar 1997 08:24:18 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=TRSystems%l=WEBSERVER2-970320132418Z-124@www.trsystems.com>
Rendered body type: text/plain

On Mar 19 David Willmore wrote:
Hey, could you let me know what you're final disposition is on this? I'm
looking at writing a generic C version of such a beast for use in a
public domain (well, GPL) SSTV decoder. It's not intended to be
realtime, so the amount of processing time taken by a generic CPU isn't
a factor. If you want more info, feel free to let me know.

Cheers,
David

>--------
I don't think a final ultimate solution was discussed. Someone posted
that they had success with an arctan(Q/I) approximation detector. The
most complete response came from Robert McGwier back in February who
provided an almost "cookbook" explanation of a complex PLL detector
using matched Hilbert bandpass filters to obtain I and Q.
I managed to implement his algorithm on the DSP-93 box with pretty good
results.(thanks Robert!) My roadblock now is finding a front end SSTV
software package that likes incoming data in async serial format. (I
really don't want to spend the rest of my life writing a Windoze SSTV
user interface!) The only program I found was JVFAX but it has problems
with Hsync even on noise free data. Everybody else is using the simple
"Hamcomm" type comparator or a soundcard as input. Any suggestions here
would be welcome.
If you are interested in any more details drop me a line and we can
discuss "off SIG"

73's
Moe Wheatley AE4JY
mwheatle@trsystems.com

13. Re: [HFSIG:2034] Re: Decisions, decisions

From: Phil Karn <karn@laptop.ka9q.ampr.org>
Date: Thu, 20 Mar 1997 21:33:21 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <m0w7wx7-000RzyC@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>This is NOT correct. It computes the maximum likelihood path as you
>said
>and it is NOT the same path as minimizing the least square error. The
>minimum bit error rate is achieved by computing the symbol at each time
>that gives the maximum likelihood given ALL of the observations, i.e.
>you
>do a posteriori estimation. In a linear channel with Gaussian noise,
>THIS
>minimizes least square error. It is terribly confusing until you pour
>through all of the math associated with it.

Yes, I should have qualified my statement by saying that it assumed an
additive gaussian white noise channel. So much of the theory assumes that,
not because it's accurate but because it makes the analysis tractable...

Phil

14. RE: SSTV Demod

From: "Wheatley, Maurice" <mwheatle@trsystems.com>
Date: Fri, 21 Mar 1997 08:27:30 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=TRSystems%l=WEBSERVER2-970321132730Z-257@www.trsystems.com>
Rendered body type: text/plain

>Hi Moe,

>I cannot report the same success with the algorithm as given, but perhaps
>you have had further insight that I did not. In particular, I don't see how
>the phase updating can work:

>   p=p + F + a*phase_err

>For the values that I have chosen, phase_err is usually small, F stays near
>the center frequency, thus p blows up (diverges) real fast, typically after
>a couple of samples. Is there some magic value for "a".

>What's the story?

>73's Johan, KC7WW

Johan,
Maybe I just got lucky but the PLL described by Robert McGwier seems to work OK. Maybe
the confusion is in the terminology. In Robert's equation (p=p + F + a*phase_err), the p is the
phase accumulator of the vco and is a modulus counter that rolls over every 2PI. It is the index
into the sin/cos lookup table. F is the vco phase increment value or frequency. When locked,
the phase error is constant, F is constant, and the phase accumulator advances at a constant
rate equal to the input frequency.

Here is my interpretation of his design as snipped from my DSP-93 pseudo code comments.

```
;=================================================================
;      =======   P r o c e s s I n p u t ( input )     =======
;=================================================================
;This routine is called for every A/D sample and processes the AGC'ed input
;sample.  The input is bandpass filtered into two channels(I andQ) using,
;a normal bandpass filter for the Iterm and an identical bandpass filter with
;a hilbert transform for the Qterm.  A complex vco is implemented with a
;phase accumulator(vcophzacc) that is added to by a phase error(Phzerr) term
;and a frequency term (vcofreq).
;
;void ProcessInput( input )
;{
;   BPFdelay[ 0 ] = input;      // do FIR calculations to get I and Q
;   acc = 0;
;   for( i=BPFIR_LENGTH-1; i >= 0; i-- ){
;      acc += BPFdelay[ i ] * IBPFIRCOEF[ i ];
;   }
;   Iterm = acc;                // Iterm =  I bandpass output data
;   for( i=BPFIR_LENGTH-1; i >= 0; i-- ){
;      acc += BPFdelay[ i ] * QBPFIRCOEF[ i ];
;      BPFdelay[ i+1 ] = BPFdelay[ i ];         // do the shuffle
;   }
;   Qterm = acc;                // Qterm =  Q hilbert bandpass output data
;
;   vcophzacc = vcophzacc + vcofreq + Phzerr;   // update new phase
;   vco_sin = SIN_TABLE[ vcophzacc>>8 ];        // create VCO sin and cos
;   vco_cos = SIN_TABLE[ vcophzacc>>8 + PHASE90 ];
;
;   Lock = Iterm*vco_cos + Qterm*vco_sin;       // calc lock signal
;   Check_lock();                               // see if phaselocked
;
;   Phzerr = Qterm*vco_cos - Iterm*vco_sin;     // calc phase error
;   Phzerr = Phzerr * Phase_const;              // scale phase error
;
;   vcofreq =  vcofreq + (Phzerr * Freq_const); // calc new frequency
;                                               // for phase lock
;   if( vcofreq < MIN_freq )          // clamp new frequency within bounds
;      vcofreq =  MIN_freq;
;   else
;      if( vcofreq > MAX_freq )
;         vcofreq =  MAX_freq;
;}
;=================================================================
```

The names of variables were changed but I think this is pretty much the same
algorithm. The gain terms, Phase_const, and Freq_const were determined by
"cut and try" method using an AFSK waveform as an input and monitoring the
phase error term(Phzerr) on a scope.

I started with a very small frequency gain and increased the phase gain
until
lock was achieved without too much overshoot. The frequency gain term
was
increased to make the locktime as short as possible. There is a lot of
interaction of the two controls but it appeared that the phase gain had
a
bigger affect on loop stability while the frequency gain increased the
overall
loop gain(stiffness) of the system. The biggest problem was determining
the
locked condition. Noisy signals required testing the "Lock" term for
average
peak to peak levels as well as the DC average to determine phase lock.

If your interested I can send you the whole DSP-93 program as it stands
now to
play with. I don't want to upload it yet as it still has lot's of rough
edges :>(

73's
Moe Wheatley AE4JY
mwheatle@trsystems.com

15. Re: [HFSIG:2038] RE: SSTV Demod

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 21 Mar 1997 08:20:54 -0800
To: hfsig@tapr.org
Message-ID: <199703211619.IAA02579@PEAK.ORG>
Rendered body type: text/plain

Hi Moe,

>Johan,
>Maybe I just got lucky but the PLL described by Robert McGwier seems to
>work OK. Maybe
>the confusion is in the terminology. In Robert's equation (p=p + F +
>a*phase_err), the p is the
>phase accumulator of the vco and is a modulus counter that rolls over
>every 2PI. It is the index
>into the sin/cos lookup table. F is the vco phase increment value or
>frequency. When locked,
>the phase error is constant, F is constant, and the phase accumulator
>advances at a constant
>rate equal to the input frequency.
>

Thanks much for the explanations - I did some further work on it yesterday
and the units of the variables became evident and now confirmed my suspicions.

I eventually developed another FM discriminator from the Frerking book whose
performance I like very much. It does not need any tuning parameters nor
need AGC - unfortunately needs an ATAN, but fortunately is no problem for
the 56002 or 2181. Bob seem to have been stuck with the older type of DSP
chips so long that he has not yet tasted the fruits yet of working with a 66
MHz 56002 or 33 MIPS 21xx, thus what he considers as outrageous cycle usage
for algorithms, is quite manageble and done as a matter of fact now.

I do wonder now how well these FM discriminator schemes compare. Frerking
(p.256) points out that FM discrimiators need 10 dB S/N to overcome
threshold effects. This obviously is not intended for weak signals.

Johan

P.S. Any code snips are welcome here; I am not much bothered with
aesthetics, but rather interested in ideas and in learing something new.

16. Re: [HFSIG:2038] RE: SSTV Demod

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sat, 22 Mar 1997 05:25:12 -0500
To: hfsig@tapr.org
Message-ID: <3333B388.11B2@worldnet.att.net>
Rendered body type: text/plain

Johan wrote:
>

> >Hi Moe,
>
> >I cannot report the same success with the algorithm as given, but perhaps
>
> >73's Johan, KC7WW
>
> Johan,
>> a*phase_err), the p is the

Moe replied:

> phase accumulator of the vco and is a modulus counter that rolls over
> every 2PI. It is the index
> into the sin/cos lookup table. F is the vco phase increment value or
> frequency. When locked,
> the phase error is constant, F is constant, and the phase accumulator
> advances at a constant
> rate equal to the input frequency.
>
> Here is my interpretation of his design as snipped from my DSP-93 pseudo
> code comments.
>
> ;
> ;   vcophzacc = vcophzacc + vcofreq + Phzerr;   // update new phase
> ;   vco_sin = SIN_TABLE[vcophzacc>>8];   // create VCO sin and
> cos
> ;   vco_cos = SIN_TABLE[vcophzacc>>8 + PHASE90];
> ;

> 73's
> Moe Wheatley AE4JY
> mwheatle@trsystems.com

Moe:

Johan has correctly pointed out that I forgot to do modulo 2pi (I always
use a quantization of [0,2pi) that is a power of two then modulo
arithmetic
is simply doing a mask with 2^n -1 where n is the power of 2 you want.

I do not see where you are doing the mask, Moe. I take it that you are
using the 16 bit number and letting the overflow die and then
using shifting right by 8. THIS WILL NOT WORK ON THE DSP5600X. It
is the reason Johan is having a problem. Johan, if you are going to
use the on chip sin table, it is size 2^n, when you compute the phase
update, simply mask by 2^n-1 and then you will stay inside the table.

One other comment:

>   Phzerr = Phzerr * Phase_const;   // scale phase error
>;
>;   vcofreq = vcofreq + (Phzerr * Freq_const); // calc new frequency
>;

if you use Freq_const = 0.25*Phase_const

then you are critically damped. I said Freq_const = 0.25*Phase_const^2
but in the implementation I gave, I did not cascade the multiplies the
way you have done. They will both work. Try putting the computation

Freq_const = 0.25*Phase_const

in the initialization of the routine and then play with Phase_const
until you achieve the locking range you want. I bet you do not have
to change Phase_const much at all.

Bob

--
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
*

```
* (H) 609-443-8963         * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061         * Buffalo (NE-III-120).  George Washington
*
* (W) 609-279-6240         * Council.  Brownie Troop 193
*
************************************************************************
```

17. Re: [HFSIG:2040] RE: SSTV Demod

```
From: forrerj@peak.org (Johan Forrer)
Date: Sun, 23 Mar 1997 11:21:34 -0800
To: hfsig@tapr.org
Message-ID: <199703231920.LAA19034@PEAK.ORG>
Rendered body type: text/plain
```

Bob,

I follow the rationale now - thanks. Your scheme is simple enough; however though, it is extremely difficult to build a suitably reliable audio AGC for operation on HF - I have seen many attempts at this. This perhaps is the weakness here.

>From what I can see, this scheme will be comparable in performance to a PLL, i.e., minimum S/N 8-9dB. However, when dealing with narrow-band non-optimal modulation formats, we have to to the best we can.

Johan, KC7WW

18. TAPR Board Election -- Don't forget to vote!

```
From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Sun, 23 Mar 1997 14:05:32 -0600
To: "APRS SIG  list   mailing"<aprssig@tapr.org>,
        " DSP-93 Build "<dsp-93@tapr.org>,
        "HF SIG  list   mailing"<hfsig@tapr.org>,
        "NETSIG  list   mailing"<netsig@tapr.org>,
        "BBS SIG  list   mailing"<bbssig@tapr.org>,
        " TAPR/AMSAT DSP "<dsp@tapr.org>,
        "TAPR-BB  list   mailing"<tapr-bb@tapr.org>,
        TAPR Regional Freq <regional_freq@tapr.org>,
        " tapr-tnc "<tapr-tnc@tapr.org>, " Spread Spectrum "<ss@tapr.org>
Message-ID: <v03007814af5b3a76bea5@[208.134.134.40]>
Rendered body type: text/plain
```

TAPR Board of Directors Elections

If you are a member of TAPR, don't forget to take the time to vote!

We have four people running for three places, so it could be your vote that determines who is on the TAPR Board of Directors.

Deadline for balloting is March 30th, 1997.

This year, members of TAPR can vote either by mail in ballot or over the Internet using the world wide web. Check your PSR on how to submit your vote via the web.

Don't lose your PSR, because you will need information contained on your mail label. If you have problems using the web system, contact Dorothy at the office and we can look into the issue.

PSRs were mailed on March 5th, so all members should have them by now. If you have not received your PSR, call or send e-mail to the office and request a ballot be sent or request the necessary information to allow you to ballot using the on-line Web page.

**** DEADLINE March 30th! ****

Tucson Amateur Packet Radio
8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399 * 817-383-0000
---
e-mail: TAPR@TAPR.ORG ftp: ftp.tapr.org   web: <http://www.tapr.org/>
---

19. Re: [HFSIG:2031] HF antenna for apartment

From: k5cnf1@juno.com (Richard B Morrow)
Date: Sun, 23 Mar 1997 16:52:26 PST
To: hfsig@tapr.org
Message-ID: <19970323.171914.2063.6.k5cnf1@juno.com>
Rendered body type: text/plain

what you use for an antenna depends of course on what frequencies you
use.
The best bet for 10-30 Mhz is the loop, like the one that MFJ sells and
like the one that the now defunct AEA also sold. particularly if you
have to remain in a stealth mode. These antennas can be kept in the
closet or behind a couch or something until you need to use them. They
do need to be as far as yu can get them from buildings and other
conductors. Also using them in the vertical plane sitting on the ground
will usually give the best results, not horizontally, as then the height
above the ground will determine how well the antenna gets out. These
antennas are very quiet as far as static and most other noise is
concerned.

hope this helps

73,
K5CNF

20. Re: [HFSIG:2043] Re: HF antenna for apartment

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Sun, 23 Mar 1997 17:44:14 -0600
To: hfsig@tapr.org
Message-ID: <v0300781daf5b709f7c58@[208.134.134.40]>
Rendered body type: text/plain

>what you use for an antenna depends of course on what frequencies you
>use.
>The best bet for 10-30 Mhz is the loop, like the one that MFJ sells and
>like the one that the now defunct AEA also sold. particularly if you
>have to remain in a stealth mode. These antennas can be kept in the
>closet or behind a couch or something until you need to use them. They
>do need to be as far as yu can get them from buildings and other
>conductors. Also using them in the vertical plane sitting on the ground
>will usually give the best results, not horizontally, as then the height
>above the ground will determine how well the antenna gets out. These
>antennas are very quiet as far as static and most other noise is
>concerned.
>

If you have not heard, it was reported on Newsline this weekend that AEA
has sold its antenna product line (antennas, analyzers, and testers) to
Tempo Research of Vista, California and that Mike Lamb, N7ML, will be
taking a position within the new AEA division of Tempo Research.

Timewave of St Paul, Minn (http://www.timewave.com/) has purchased the AEA
Data Products line and some other products.

Listen to: http://www.tapr.org/newsline #1023, time segment:13:25 for more
details.

Timewave has a web page up already covering this
(http://www.timewave.com/aea.html)

Here is what that pages says:

March 17, 1997

Timewave is in the final stages of completing the purchase of the data
hardware and software product lines of AEA (Advanced Electronic
Applications, Inc.). Timewave will offer new product sales as well as
warranty service, repair, and firmware upgrades for existing AEA products.
It will be several weeks before any of the new products and services begin
to be available.

Check the Timewave Web page for further information - it will appear here
first!

Please help us by not calling via telephone until we have had a chance to
get the new product line and services organized.

Tempo Research Corp. of Vista, CA has acquired the AEA antenna analyzer and

antenna product lines. Contact them at 619-599-9677.

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd
-----

21. Re: [HFSIG:2041] RE: SSTV Demod

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sun, 23 Mar 1997 23:30:01 -0500
To: hfsig@tapr.org
Message-ID: <33360349.1FF0@worldnet.att.net>
Rendered body type: text/plain

Johan Forrer wrote:
>
> Bob,
>
> I follow the rationale now - thanks. Your scheme is simple enough; however
> though, it is extremely difficult to build a suitably reliable audio AGC for
> operation on HF - I have seen many attempts at this. This perhaps is the
> weakness here.
>
> >From what I can see, this scheme will be comparable in performance to a PLL,
> i.e., minimum S/N 8-9dB. However, when dealing with narrow-band non-optimal
> modulation formats, we have to to the best we can.
>
> Johan, KC7WW

Ahh yes. I should publish that here as well. All of my agc's use past
and
future as they should on HF and seem to keep the signals above a
threshold
well within the amplitudes needed for proper operation of the Complex
PLL (you have correctly identified it as a PLL but with all complex
arithmetic and no sum product ripples, only the lovely difference
terms).
I use thresholded agc that looks ahead a few dozen samples as well as
learning from the past. This seems to be extremely robust if expensive,
but it seems to be worth it.

As far as performance, I agree with you. 8-9 db S/N in the desired
bandwidth for tracking (say) SSTV is required.

Bob

--
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

22. Re: HF antenna for apartment

From: k5cnf1@juno.com (Richard B Morrow)
Date: Sun, 23 Mar 1997 23:17:29 PST
To: hfsig@tapr.org
Message-ID: <19970323.232754.4999.2.k5cnf1@juno.com>
Rendered body type: text/plain

Thanks for the info on AEA being sold, I had not heard that.

73,
K5CNF

23. AGC

From: "Wheatley, Maurice" <mwheatle@trsystems.com>
Date: Mon, 24 Mar 1997 12:09:24 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=TRSystems%l=WEBSERVER2-970324170924Z-10@www.trsystems.com>
Rendered body type: text/plain

Bob wrote,

>Ahh yes. I should publish that here as well. All of my agc's use past and
future as they should on HF and seem to keep >the signals above a
threshold well within the amplitudes needed for proper operation of the
Complex PLL (you have >correctly identified it as a PLL but with all
complex arithmetic and no sum product ripples, only the lovely
difference >terms). I use thresholded agc that looks ahead a few dozen
samples as well as learning from the past. This seems to be >extremely
robust if expensive, but it seems to be worth it.
>As far as performance, I agree with you. 8-9 db S/N in the desired bandwidth
for tracking (say) SSTV is required.

>Bob

I would be very interested in a discussion of AGC techniques and
requirements. Here are a few of my questions:

1. What are the requirements for an affective AGC on HF? Dynamic range
of operation? Attack and Decay times?

2. Does one use signal peaks or average signal values to calculate the
AGC value?

3. Are there any clever algorithms for determining signal peak values?
I presently am taking the derivative of the incoming signal and when the
sign changes(ie. second derivative==0) the absolute value of the last
sample is stored as the peak waveform value. These peak values are then
LP filtered to get a reasonable value for the incoming signal
amplitude. This works ok on noise free signals but I'm not sure if it
is optimum for noisy inputs.

Thanks for any ideas,

Moe

24. modem design question?

From: sscalsk@mail.ameritel.net (Stan Scalsky)
Date: Wed, 26 Mar 1997 22:16:11 +0500
To: hfsig@tapr.org
Message-ID: <2.2.16.19970326171611.37e7519c@mail.ameritel.net>
Rendered body type: text/plain

Most of the data transmission systems I am familiar with on HF use
tones that are shifted by some fixed amount but lately I have come
across systems that insert a gap in the tone arrangement. For example
the Chinese (most likely) have an 8 tone MFSK system with 4 tones shifted
by 300Hz, a 450Hz gap, 4 tones shift by 300Hz. All the tones seem to be
modulated the same so is there a purpose in modem design to inserting a
gap in the tone arrangement?

Another odd example is a system commonly found in Europe composed of 7 "tones"
at 500/800/1100/1400/2000/2600/2900 Hz. Notice the 1400/2000/2600 set shifted
by 600Hz with all the others shifted by 300Hz? What purpose would staggering
the tones serve? I suppose it could be simply a missing tone(s) but they have
never been observed.

Any ideas? Most of the texts I have been thru cover 1/T tone spacing for non-coherent
detection, etc, etc but nothing like the above.

-= stan

25. Re: [HFSIG:2048] modem design question?

From: Barry McLarnon VE3JF <bm@lynx.ve3jf.ampr.org>
Date: Sat, 29 Mar 1997 03:01:28 +0000 (GMT)
To: hfsig@tapr.org
Message-ID: <199703290301.DAA14689@lynx.ve3jf.ampr.org>
Rendered body type: text/plain

sscalsk@mail.ameritel.net (Stan Scalsky) wrote:
> Most of the data transmission systems I am familiar with on HF use

> tones that are shifted by some fixed amount but lately I have come
> across systems that insert a gap in the tone arrangement. For example
> the Chinese (most likely) have an 8 tone MFSK system with 4 tones shifted
> by 300Hz, a 450Hz gap, 4 tones shift by 300Hz. All the tones seem to be
> modulated the same so is there a purpose in modem design to inserting a
> gap in the tone arrangement?

This sounds like an in-band dual-frequency-diversity configuration - each
tone in the lower group is paired with the corresponding tone in the upper
group and carries the same information, and the demodulator applies some
type of diversity combining to each pair. The gap increases the spacing
between the diversity pairs, and hence the probability that the fading on
them will be uncorrelated (for max diversity gain).

> Another odd example is a system commonly found in Europe composed of 7 "tones"
> at 500/800/1100/1400/2000/2600/2900 Hz. Notice the 1400/2000/2600 set shifted
> by 600Hz with all the others shifted by 300Hz? What purpose would staggering
> the tones serve? I suppose it could be simply a missing tone(s) but they have
> never been observed.

Hmmm... dunno 'bout that one. It appears to be the same configuration as the
other one, but with one tone (2300 Hz) missing.

Barry

--
  Barry McLarnon VE3JF/VA3TCP | Internet: bm@hydra.carleton.ca
  Ottawa Amateur Radio Club | AMPRnet: bm@lynx.ve3jf.ampr.org
  Packet Working Group | Web: http://hydra.carleton.ca

26. Software, etc.

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sat, 29 Mar 1997 18:34:19 -0500
To: hfsig@tapr.org, dsp@tapr.org
Message-ID: <333DA6FB.39A1@worldnet.att.net>
Rendered body type: text/plain

I am sorry about the delay on the software things from me. I just
returned
from five weeks in the middle east. I will be going to Germany in May,
Rome shortly after and some odd places after that. I will do all I can
to get this stuff out during the month of April!

Bob

--
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
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* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
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* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

27. Re: [HFSIG:2048] modem design question?

From: forrerj@peak.org (Johan Forrer)
Date: Sat, 29 Mar 1997 21:15:36 -0800
To: hfsig@tapr.org
Message-ID: <199703300513.VAA01001@PEAK.ORG>
Rendered body type: text/plain

Stan,

On which frequencies did you copy these signals? I would be interested in
looking at that myself.

73, Johan KC7WW

>Most of the data transmission systems I am familiar with on HF use
>tones that are shifted by some fixed amount but lately I have come

>across systems that insert a gap in the tone arrangement. For example
>the Chinese (most likely) have an 8 tone MFSK system with 4 tones shifted
>by 300Hz, a 450Hz gap, 4 tones shift by 300Hz. All the tones seem to be
>modulated the same so is there a purpose in modem design to inserting a
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>the tones serve? I suppose it could be simply a missing tone(s) but they have
>never been observed.
>
>Any ideas? Most of the texts I have been thru cover 1/T tone spacing for
non-coherent
>detection, etc, etc but nothing like the above.
>
>-= stan
>
>

April 1997

Original source file: hfsig.9704
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9704

1. Future HF modem standard for amateur radio

From: claude@bauv106.bauv.unibw-muenchen.de (Claude Frantz)
Date: Tue, 1 Apr 1997 21:02:35 +0200 (MET DST)
To: hfsig%tapr.org@netwman.RZ.UniBw-Muenchen.de
Message-ID: <199704011902.VAA18519@bauv106.bauv.unibw-muenchen.de>
Rendered body type: text/plain

In my opinion, it's time now, to define a standard for a future HF modem for amateur radio, which can be proposed to the administration in order to avoid the reproach of using a non open coded modulation. We need a high performance, open standard designed to have good performance in the specific amateur radio environment on HF. Of course, the standard should be verified and commented in the community.

At first, the objectives have to be defined. Please allow me to start the subject, please comment.

The question of the bandwidth is open, it should be compatible with any common CW/SSB transceiver on the market. The maximum speed should be defined too. I think, there should be two modes: a point-to-point mode for classical QSOs and a broadcast mode. The question of the right modulation will probably be a very controversial subject, strongly related to the question of the bandwidth. The standard should be so that the modem can be build with commonly available up to day one-chip microprocessors together with a commonly available memory, at a reasonable cost. The reliability and speed in an enviromnent in which one the QRM and QSB can be strong over a relatively long period of time should be seen a the primary design goal. I think that procedures which try to switch between clear and noisy modes do have inherent problems related to the fact that it is very difficult to signal the mode switching to the peer when the performance of the communication channel has degraded. Another important question is related to the automatic repeat request error control scheme and forward-error-correction scheme. Remember, two modes are needed. These procedures are of great importance in respect of performance and speed.

73,
Claude

---

2. Re: [HFSIG:2052] Future HF modem standard for amateur radio

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Tue, 1 Apr 1997 14:18:36 -0600
To: hfsig@tapr.org
Message-ID: <v0300780daf6719183e75@[208.134.134.40]>
Rendered body type: text/plain

Hello Claude:

If we are talking strictly about the rules (and not technical progress), there is no need to further define operations within the rules.

If anything, we need to be asking the FCC to eliminate wording in the rules to foster such experimentation instead of asking for changes to define yet another niche that might never happen or by the time the rules are finally confirmed (a 2-5 year process) we have already left behind.

As of right now there is no modulation or coding issues limiting HF operations. Well, there are a few, but not in the sense of the context below. The FCC stated four years ago that it was enough to have whatever was being done published in the Digital Communications Conference or other recognized publication that could be referenced. I would have to go look, but the Future Systems Committee of the ARRL requested and got changed the rules that allow new modes to be defined in this manner, instead of having them submitted for rules inclusion. That was one reason we saw Clover, PACTOR and others published in the #11 DCC (http://www.tapr.org/tapr/html/cnc11.html) and others in 13th DCC. All the abstracts can be found at http://www.tapr.org/tapr/html/publications.html.

As to a technical standard -- past history indicates that standards from within the ham community before something exists, while providing an excellent intellectual exercise, rarely influences the eventual outcome of

a design by another ham(s). Maybe someone can point me to something within
the amateur community that actually is an example of this working before
something was coded or prototyped.

There have been a lot of discussion at various HF SIG meetings and most it
comes down to the fact that we need rules within Part 97 to allow us to
really get back to experimenting with new modes and moving the art forward
instead of being hackled by outdated rules that are in place to limit our
impact upon a community that wants good, simple, cheap communications
instead of technical expertise and evolution.

If anything, let's focus on getting the rules rewritten so that there are
no limits to what we can experiment with.

Cheers - Greg, WD5IVD

>In my opinion, it's time now, to define a standard for a future HF
>modem for amateur radio, which can be proposed to the administration
>in order to avoid the reproach of using a non open coded modulation.
>We need a high performance, open standard designed to have good
>performance in the specific amateur radio environment on HF. Of
>course, the standard should be verified and commented in the
>community.
>
>At first, the objectives have to be defined. Please allow me to
>start the subject, please comment.
>
>The question of the bandwidth is open, it should be compatible with
>any common CW/SSB transceiver on the market. The maximum speed
>should be defined too. I think, there should be two modes: a
>point-to-point mode for classical QSOs and a broadcast mode. The
>question of the right modulation will probably be a very
>controversial subject, strongly related to the question of the
>bandwidth. The standard should be so that the modem can be build
>with commonly available up to day one-chip microprocessors together
>with a commonly available memory, at a reasonable cost. The
>reliability and speed in an enviromnent in which one the QRM and QSB
>can be strong over a relatively long period of time should be seen a
>the primary design goal. I think that procedures which try to switch
>between clear and noisy modes do have inherent problems related to
>the fact that it is very difficult to signal the mode switching to
>the peer when the performance of the communication channel has
>degraded. Another important question is related to the automatic
>repeat request error control scheme and forward-error-correction
>scheme. Remember, two modes are needed. These procedures are of
>great importance in respect of performance and speed.
>
>73,
>Claude

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd
-----

3. Re: [HFSIG:2052] HFSIG digest 297/reply#1

From: "JerryL." <glockett@lightspeed.net>
Date: Tue, 01 Apr 1997 13:05:37 -0800
To: hfsig@tapr.org
Message-ID: <334178A1.4B5B@lightspeed.net>
Rendered body type: text/plain

hfsig@tapr.org wrote:
>
>   HFSIG Digest 297
>
> Topics covered in this issue include:
>
>   1) Future HF modem standard for amateur radio
>   by claude@bauv106.bauv.unibw-muenchen.de (Claude Frantz)
>
> --
>   <snip>
>
> At first, the objectives have to be defined. Please allow me to
> start the subject, please comment.

> <snip>
> repeat request error control scheme and forward-error-correction
> scheme. Remember, two modes are needed. These procedures are of
> great importance in respect of performance and speed.
>
> 73,

Claude,

It seems to me, that one of the real 'first' things to is to define a protocol that allows the maximum use of technology while at the same time keeping a open architecture such that 'everyone' can monitor all the conversations they can hear. To many of these schemes, while technically superior exclude the old 'roundtable' of years past, ala AM, SSB, and Rtty. I always wondered why Fec didn't catch on better over Amtor, as I usually got very good copy from the ARRL broadcasts. All you had to do was tune them in, while Amtor and all the variations that have come along are hard to copy unless you are one of the two original stations in the QSO. Now with SS and Clover (yes I have Clover and enjoy it and the other modes) it is getting closer and closer to a 'closed shop' for a lot of people. Frankly, I have almost every mode except Pactor ll, and SS, and I find myself still using that archaic mode
of Rtty, simply because its easy to monitor everyone and join in when the mood strikes. A lot of times I for one, don't feel comfortable in joining in unless I have listened for some period of time. Using trick coding systems is great, but it would be nice if there were one good standard defined that would cover both areas. Some state of the art tradeoffs are inevitable but it seems to me if we don't cover this ground at some point we are all going to be doing our own thing while the whole hobby sorta grinds down from the lack of general monitoring that leads to all sorts of interesting input (or output!) from people...

Now,come on flames <grin>!

de Jerry (wb6wpx)

A Serious Collins/BA/Drake Enthusiast ! (and some solid state!)

glockett@lightspeed.net
Phone/Fax (24-hrs) 805-589-4120

4. Re: [HFSIG:2052] Future HF modem standard for amateur radio

From: Bill Carver <bcarver@magiclink.com>
Date: Tue, 1 Apr 1997 15:58:43 -0600 (CST)
To: hfsig@tapr.org, hfsig@tapr.org
Message-ID: <199704012158.PAA15114@tapr.org>
Rendered body type: text/plain

I agree with Claude.

To expand on what he said, it seems to me that commercial vendors have championed their own (incompatible) digital modes. Fine for them (if theirs becomes accepted), but not good for us. The cohesion of digital operation with a common set of standards (unlike the olden days of 45 baud RTTY) has been lost.

TAPR has an opportunity to champion a rational, technically competent HF modem, and thereby strengthen amateur digital communication techniques and interoperability.

High speed point-to-point digital communications is great for commercial users. But except for perhaps a few traffic operations, does that serve amateur needs? It makes it impossible to "read the mail", making digital a more closed group, difficult to recruit and interest new users. I am much more concerned with the ability of a geographically diverse group with common interests to share a frequency and reliably monitor 100 wpm traffic than I am being able to send files point-to-point at some much higher rate.

Regards - Bill - W7AAZ

5. Re: [HFSIG:2055] Re: Future HF modem standard for amateur radio

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Tue, 1 Apr 1997 17:07:05 -0600
To: hfsig@tapr.org
Message-ID: <v0300781daf673e0aef5c@[208.134.134.40]>
Rendered body type: text/plain

Before we run off and reinvent the wheel, has everyone gone and listened to Johan's presentation at last year's HF SIG meeting as well as the HF SIG meeting as a whole ?

http://www.tapr.org/tapr/html/dcc96audio.html#hfsig

Specifically the one on ' Quadtor Development Update '

Agreed that this is a chance for the HF digital community and maybe TAPR to do something (which has been discussed in past threads as well for two years now); however, history shows that this stuff doesn't get done by committee or large e-mail lists. TAPR doesn't form a group and then it happens. 95% of the time some person or group approaches TAPR with a concept or working unit and say's "what you think -- a kit or something that can help the amateur community ?" Then a small support group might get formed to make final things happen to a design.

Someone will have to develop something and then make it available via TAPR or whatever means in an open format so that others can use it and expand upon it -- that will accomplish what you want. Although the effort and time required to do this sometimes makes it hard for many to ignore the potential commercial possibilities for original work like this. Why give it away, unless you are already well off, when you can make some money commercially.

What most don't figure out is you can't make the money in the amateur market. That is where most hams miss the boat. They have something neat, but instead of providing it to the amateur community and nonprofit educational concerns and the like for free or some small fee and selling the commercial rights, they try to just do the amateur market. Big mistake. Give the amateur rights to TAPR or whoever to make kits available and take a royalty and then either find an agent to handle the commercial side of things or do it yourself. By trying to make money in the amateur market, you end up jacking up the cost so that it excludes a lot of folks....I'll get of my soap box now.

TAPR sponsors many lists and for the most part, while this one is an exception, most of them have talkers and discussers and very few people implementing. Don't get me wrong, discussion, inclusion, and information sharing is what having a SIG e-mail list is all about. The SIGs are suppose to be gathering points for common interests -- there goal is not to be R&D incubators. To emphasize this, the track record of new kits coming from these discussions list is zlitch. These lists typically provide a conduit and educational means for sharing information and knowledge. I am hoping that someone will eventually prove me wrong though. That a list will get enough focus to really produce something. If anything, what happens is that these lists produce a ciritcal mass of interest someplace else that goes off and does something off the list. That is what happens more often than not.

There are plenty of standards to chose from that meet what folks want with a little modification. Chose one and start developing. Then tell everyone about it and put it out so others can expand upon it. Starting by writing a spec is probably the least productive way to get things off the ground...at least taking a lot of time to write one in a large group. Writing your own spec for development is one thing, writing one by a list committee is another. The best approach is to write your completed spec and then ask for feedback and then start implementing.

This is one reason we see within the amateur community the same folks showing up with something in hand to implement and hand off for development. It takes a lot of energy, focus, and gumption to stick begin and follow through with a project without distractions for 6-18 months or more till it is completed --- all in your spare time with family, wife, animals, and the like ever present in the background.

Just a few of my thoughts.

Discussion is good, but someone has to implement eventually. Who is ready to step forward and make it happen ?

Cheers - Greg

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org

http://www.tapr.org/~wd5ivd
-----

6. Re: [HFSIG:2053] Re: Future HF modem standard for amateur radio

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 01 Apr 1997 15:45:37 -0800
To: hfsig@tapr.org
Message-ID: <199704012342.PAA29799@PEAK.ORG>
Rendered body type: text/plain

>Hello Claude:
>
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>If anything, we need to be asking the FCC to eliminate wording in the rules
>to foster such experimentation instead of asking for changes to define yet
>another niche that might never happen or by the time the rules are finally
>confirmed (a 2-5 year process) we have already left behind.

Claude, Greg,

Considering the state of amateur HF digital development, discussions on this topic is most welcome; let's go for it! Both on legalese and technical aspects.

I think we, as a discussion group, have visited and revisited many of the essentials of such a modern HF digital system over the past few years. There are indeed a number of fun things to pursue and a lot to be learned along the way - it is not a trivial undertaking to say the least. Perhaps it is time to take fresh look at things.

Johan, KC7WW

>
>As of right now there is no modulation or coding issues limiting HF
>operations. Well, there are a few, but not in the sense of the context
>below. The FCC stated four years ago that it was enough to have whatever
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>PACTOR and others published in the #11 DCC
>(http://www.tapr.org/tapr/html/cnc11.html) and others in 13th DCC. All the
>abstracts can be found at http://www.tapr.org/tapr/html/publications.html.
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>within the ham community before something exists, while providing an
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>a design by another ham(s). Maybe someone can point me to something within
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>>repeat request error control scheme and forward-error-correction
>>scheme. Remember, two modes are needed. These procedures are of
>>great importance in respect of performance and speed.
>>
>>73,
>>Claude
>
>
>-----
>Greg Jones, WD5IVD
>Austin, Texas
>wd5ivd@tapr.org
>http://www.tapr.org/~wd5ivd
>-----
>
>
>

7. Re: [HFSIG:2053] Re: Future HF modem standard for amateur radio

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Wed, 02 Apr 1997 09:22:45 +0200
To: hfsig@tapr.org
Message-ID: <33420945.27E3@ife.ee.ethz.ch>
Rendered body type: text/plain

> below. The FCC stated four years ago that it was enough to have whatever
> was being done published in the Digital Communications Conference or other
> recognized publication that could be referenced. I would have to go look,

Maybe I'm missing something, but as far as I know
neither Pactor 2 nor Clover were documented to the extent
that someone else could implement at least a monitoring
device. The articles I've seen only contained 'technobabble'.
I'd hope that this wasn't enough for the FCC and other
authorities, but I assume this is not the case...

73s
Tom
--
Thomas (Tom) Sailer
Weinbergstrasse 76
CH-8408 Winterthur

EMail: sailer@ife.ee.ethz.ch
Ham Radio: hb9jnx @ hb9w.che.eu
Phone: ++41 52 222 32 81

8. Re: [HFSIG:2056] Re: Future HF modem standard for amateur radio

From: Claude Frantz <claude@bauv106.bauv.unibw-muenchen.de>
Date: Wed, 2 Apr 1997 10:53:18 +0200 (MET DST)
To: hfsig@tapr.org@netwman.RZ.UniBw-Muenchen.de
Message-ID: <199704020853.KAA18898@bauv106.bauv.unibw-muenchen.de>
Rendered body type: text/plain

According to Greg Jones, WD5IVD:

Greg,

I share most of your opinion in your message, but allow me some comments.

> Agreed that this is a chance for the HF digital community and maybe TAPR to
> do something (which has been discussed in past threads as well for two
> years now); however, history shows that this stuff doesn't get done by
> committee or large e-mail lists.

This is probably right but there is no reason to continue this way.

In the past it was often possible to use not-so-specific equipment
in the amateur radio area but with the increasing complexity of the
used systems, this will not more be possible. The incompatibily will
to total. There is no real problem to use a commercial TU to copy and
send amateur RTTY.

> 95% of the time some person or group approaches TAPR with a
> concept or working unit and say's "what you think -- a kit or something
> that can help the amateur community ?" Then a small support group might
> get formed to make final things happen to a design.

Due to the complex matter, I understand well this request. If a standard
will be adopted, there should be kits available and some other tools too,
in order to help every ham to join the party. The DSP-93 project is a
good example.

> There are plenty of standards to chose from that meet what folks want with
> a little modification. Chose one and start developing. Then tell everyone
> about it and put it out so others can expand upon it. Starting by writing
> a spec is probably the least productive way to get things off the
> ground...at least taking a lot of time to write one in a large group.
> Writing your own spec for development is one thing, writing one by a list
> committee is another. The best approach is to write your completed spec
> and then ask for feedback and then start implementing.

You are probably right here too, but the matter has become so complex
that one ham alone can hardly have the necessary skill to design the
whole device. In the commercial area this not different, there are
design teams and seldom single designers. Further, there is still the
problem that a single person considers only the problem from a more or
less personal point of view which is probably not shared by the whole
community. A standard has to be useful for the whole ham community.

I think that the theoretical work of this SIG should not remain only
theoretical, but should result in a practical work. I will probably
not be the right goal to speak and write about all this matter and
to use a Clover, Pactor, xxxtor device in the QSOs.

73,
Claude
DJ0OT

9. Re: [HFSIG:2058] Re: Future HF modem standard for amateur radio

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Wed, 2 Apr 1997 13:40:59 -0600
To: hfsig@tapr.org
Message-ID: <v03007802af6860a9ac15@[208.134.134.40]>
Rendered body type: text/plain

>> below. The FCC stated four years ago that it was enough to have whatever
>> was being done published in the Digital Communications Conference or other
>> recognized publication that could be referenced. I would have to go look,
>
>Maybe I'm missing something, but as far as I know
>neither Pactor 2 nor Clover were documented to the extent
>that someone else could implement at least a monitoring
>device. The articles I've seen only contained 'technobabble'.
>I'd hope that this wasn't enough for the FCC and other
>authorities, but I assume this is not the case...
>

I agree that the depth of information published was not enough to design
your own system without spending some major reverse engineering effort, but
you would have to talk to Paul Rinaldo, W4RI@ARRL.ORG about this point.

Paul works at the ARRL DC office and was the major driver behind getting
the rule set so that we could easily add new data techniques without be
limited to what was published in the rules. Personally, this was really
needed, if you consider at the time all the digital modes except packet and
RTTY were being used against the rules, without this change.

Let's not worry about what others have published. None of the stuff
published thus far was meant to be used by amateurs other than as a
commercial product we purchased. Amateur radio is reverting back some to
its experimental ways as more and more of these companies close their
doors. The important part to remember is that we can implement something
and then make as much of it published as you want in order to get it
accepted as a digital mode.

We has discussed the level of rules issue over on the SS list a lot. When
we change the rules, we want to make them simpler to promote communications
and experimentation. I am glad the rules don't specify the level of
information release you have to make available on a digital code, since we
as amateurs should make that decision, not Part 97. Part 97 has enough
silly stuff in it that places major hackles on experimentation. Granted
that the commercial companies involved in the codes published thus far
didn't make a lot of it available, but that doesn't mean that we have to
follow the same amount of disclosure.

The section of the rules is: 97.309(a)4.
An amateur station transmitting a RTTY or data emission using a digital
code specified in this paragraph may use technique whose technical
characteristics have been documented publicly, such as CLOVER, G-TOR, or
PacTOR, for the purpose of facilitating communications.

It doesn't say you have to give it away, it just says that you have to
document enough so that FCC knows what it is. The documents in the DCC do
that for them.

Now that I have quoted from the Rules, let me add -- the rules are a living
thing. Don't let what the rules say stop you from thinking about
developing something new that isn't in the rules. We can change the rules
and doubly so if the new technique is really useful.

The key is that we have to advance forward. Implement and then let others
know what you have so that it can spread.

Cheers - Greg

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd
-----

10. Re: [HFSIG:2059] Re: Future HF modem standard for amateur radio

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Wed, 2 Apr 1997 13:43:23 -0600
To: hfsig@tapr.org
Message-ID: <v03007803af6867152ebe@[208.134.134.40]>
Rendered body type: text/plain

I agree with what you are saying Claude. Hopefully our comments have
helped other to understand the problem related to the task of writing such
documents and then trying to implement them.

Ever forward as Johan pointed out.

Cheers -- Greg

>I share most of your opinion in your message, but allow me some comments.
>
>> Agreed that this is a chance for the HF digital community and maybe TAPR to
>> do something (which has been discussed in past threads as well for two
>> years now); however, history shows that this stuff doesn't get done by
>> committee or large e-mail lists.
>
>This is probably right but there is no reason to continue this way.
>In the past it was often possible to use not-so-specific equipment
>in the amateur radio area but with the increasing complexity of the
>used systems, this will not more be possible. The incompatibily will
>to total. There is no real problem to use a commercial TU to copy and
>send amateur RTTY.
>
>> 95% of the time some person or group approaches TAPR with a
>> concept or working unit and say's "what you think -- a kit or something
>> that can help the amateur community ?" Then a small support group might
>> get formed to make final things happen to a design.
>
>Due to the complex matter, I understand well this request. If a standard
>will be adopted, there should be kits available and some other tools too,
>in order to help every ham to join the party. The DSP-93 project is a
>good example.
>
>> There are plenty of standards to chose from that meet what folks want with
>> a little modification. Chose one and start developing. Then tell everyone
>> about it and put it out so others can expand upon it. Starting by writing

>> a spec is probably the least productive way to get things off the
>> ground...at least taking a lot of time to write one in a large group.
>> Writing your own spec for development is one thing, writing one by a list
>> committee is another. The best approach is to write your completed spec
>> and then ask for feedback and then start implementing.
>
>You are probably right here too, but the matter has become so complex
>that one ham alone can hardly have the necessary skill to design the
>whole device. In the commercial area this not different, there are
>design teams and seldom single designers. Further, there is still the
>problem that a single person considers only the problem from a more or
>less personal point of view which is probably not shared by the whole
>community. A standard has to be useful for the whole ham community.
>
>I think that the theoretical work of this SIG should not remain only
>theoretical, but should result in a practical work. I will probably
>not be the right goal to speak and write about all this matter and
>to use a Clover, Pactor, xxxtor device in the QSOs.
>
>73,
>Claude
>DJ0OT

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd
-----

11. Re: [HFSIG:2060] Re: Future HF modem standard for amateur radio

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 03 Apr 1997 09:09:15 -0800
To: hfsig@tapr.org
Message-ID: <199704031706.JAA20297@PEAK.ORG>
Rendered body type: text/plain

Greg,

>Let's not worry about what others have published. None of the stuff
>published thus far was meant to be used by amateurs other than as a
>commercial product we purchased. Amateur radio is reverting back some to
>its experimental ways as more and more of these companies close their
>doors. The important part to remember is that we can implement something
>and then make as much of it published as you want in order to get it
>accepted as a digital mode.

This unfortunately is reality and I don't think there would ever be
satisfactory disclosure; On the other hand, perhaps it should perhaps be
better left alone - I recently had the opportunity to evaluate, using my HF
channel simulator, a number of commercial-grade HF modems, including some of
those mentioned along the way. It is not a pretty picture at all. With all
due respects, let me assure anyone that it is not really worth pursuing
these and expect a payback for your efforts - we can and must simply do
better! One little ray of hope is that those units based on modern
technology are indeed outstanding.

Having said that, we have to realise that there are a number of factors
playing an important role at this point in time; whether you are doing this
for a hobby or as a professional. This is how I see it:

           Technical issues:
           ----

1) The sunspot cycle - we are at a very low point. Signals are weak,
propagation is sporadic and variable; communications are stressed. There
will be an increased reliance on high tech solutions.

2) There is growing acceptance/assurance within the radio community to
allow/encourage innovative experimentation; the use of bandwidth compression
techniques, bandwith efficiency, and/or fault tolerant schemes will become a
future neccessity.

3) The use of "wider" bandwith emissions, when done with due care, is
becoming possible.

4) Respectable DSP platforms and appropiate software development tools are
becoming available to amateurs.

5) There is a great need for education, training, and awareness of the possibilities in this field. We are in urgent need for contributions and sharing of experiences on a wide range of topics.

Non-technical/political issues:
---

1) The role that HF communications plays in the amateur radio setting: can we still claim our role for emergency/community service? Has the Internet replaced this need? What is our mission: educational, experimentation, hobby?

2) The impact of the Internet has certainly affected HF digital communications; as the sole means for long distance communications; the congestion associated with the digital band segments.

My 2 cents worth.

Johan, KC7WW

12. Re: [HFSIG:2062] Re: Future HF modem standard for amateur ra

From: RLANIER@mailb.harris.com (RLANIER)
Date: Thu, 3 Apr 1997 16:13:16 -0500
To: hfsig@tapr.org
Message-ID: <3441ddc0@mailb.harris.com>
Rendered body type: text/plain

I'd like to comment on these issues:

>1) The role that HF communications plays in the amateur radio setting: can
>we still claim our role for emergency/community service? Has the Internet
>replaced this need? What is our mission: educational, experimentation, hobby?

I think all three. Amateur radio is a hobby for some, an experimental test bed for others and also an educational medium to teach young and old about wireless communications.

>2) The impact of the Internet has certainly affected HF digital
>communications; as the sole means for long distance communications; the
>congestion associated with the digital band segments.

I have heard this before and I don't agree with this. As demonstrated by some rouge experimenters, a wireless link can be used as a connection to an Internet Service Provider, allowing much faster data rates than the conventional 28.8 or 33.6 modem. We can't use the Internet to connect to a space probe orbiting Mars or talk to the astronauts aboard the space shuttle. Ther HAS to be a wireless link in there SOMEWHERE.

At the most, the Internet is a supplement to wireless communications, NOT a replacement.

>My 2 cents worth.

>Johan, KC7WW

That's my two cents ...

Tony KE4ATO

13. Re: Future HF modem standard for amateur ra

From: k5cnf1@juno.com (Richard B Morrow)
Date: Thu, 3 Apr 1997 19:55:55 PST
To: hfsig@tapr.org
Message-ID: <19970403.200301.5479.0.k5cnf1@juno.com>
Rendered body type: text/plain

we still claim our role for emergency/community service? Has the
>Internet
>>replaced this need? What is our mission: educational,
>experimentation, hobby?
>
>   I think all three. Amateur radio is a hobby for some, an
>experimental
>   test bed for others and also an educational medium to teach young
>and
>   old about wireless communications.

>
>>2) The impact of the Internet has certainly affected HF digital
>>communications; as the sole means for long distance communications;
>the
>>congestion associated with the digital band segments.

I have yet to hear of Pitcarin Island on the Internet, but got a QSL for
a ten meter contact. All of what you say is true and bears remembering
and quoting to the infidels that think wire is the only to go.
>
>   I have heard this before and I don't agree with this. As
>demonstrated
>   by some rouge experimenters, a wireless link can be used as a
>   connection to an Internet Service Provider, allowing much faster
>data
>   rates than the conventional 28.8 or 33.6 modem. We can't use the
>   Internet to connect to a space probe orbiting Mars or talk to the
>
>   astronauts aboard the space shuttle. Ther HAS to be a wireless
>link in
>   there SOMEWHERE.
>
>   At the most, the Internet is a supplement to wireless
>communications,
>   NOT a replacement.
>
>>My 2 cents worth.
>
>>Johan, KC7WW
>
>
>   That's my two cents ...
>
>   Tony KE4ATO
>
>
And mine also, I have had my license for 41 years and you would be
suprized how many so called new communications were being called the
doomsday bomb for the ham bands.

Richard, K5CNF

14. Re: [HFSIG:2058] Re: Future HF modem standard for amateur radio

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 3 Apr 1997 20:34:51 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199704040434.UAA13101@servo.qualcomm.com>
Rendered body type: text/plain

>Maybe I'm missing something, but as far as I know
>neither Pactor 2 nor Clover were documented to the extent
>that someone else could implement at least a monitoring
>device. The articles I've seen only contained 'technobabble'.

Agreed. Certainly is counter to the amateur spirit, in my opinion.

>I'd hope that this wasn't enough for the FCC and other
>authorities, but I assume this is not the case...

The FCC probably only cares that they can go out and buy one of the
units and use it to monitor. This requirement is met by all of the
commercial units.

Phil

15. Users of Motorola EVM56

From: Kjell Midtseter <kmidtset@online.no>
Date: Fri, 04 Apr 1997 08:46:24 +0100
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970404074624.008d5110@online.no>
Rendered body type: text/plain

The latest versions of the C compiler, assembler, linker and simulator
development software for the 56000 series of DSPs may be downloaded from
http://www.mot.com/sps/dsp/software/index.html

This www page also contains links to EVM28 which is the latest version 2.8
of the EVM56 software and to additional software in the new dr. Bub file.

BY selecting the Documentation ICON you may also locate manuals for the C
compiler, assembler and linker.

73s de LA3SG

16. Re: [HFSIG:2054] Re: HFSIG digest 297/reply#1

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 04 Apr 1997 08:12:07 -0800
To: hfsig@tapr.org
Message-ID: <199704041608.IAA02002@PEAK.ORG>
Rendered body type: text/plain

Jerry,

>It seems to me, that one of the real 'first' things to is to define a
>protocol that allows the maximum use of technology while at the same
>time keeping a open architecture such that 'everyone' can monitor all
>the conversations they can hear. To many of these schemes, while
>technically superior exclude the old 'roundtable' of years past, ala
>AM, SSB, and Rtty. I always wondered why Fec didn't catch on better
>over Amtor, as I usually got very good copy from the ARRL broadcasts.
>All you had to do was tune them in, while Amtor and all the variations
>that have come along are hard to copy unless you are one of the two
>original stations in the QSO. Now with SS and Clover (yes I have Clover
>and enjoy it and the other modes) it is getting closer and closer to a
>'closed shop' for a lot of people. Frankly, I have almost every mode
>except Pactor ll, and SS, and I find myself still using that archaic
>mode
>of Rtty, simply because its easy to monitor everyone and join in when
>the mood strikes. A lot of times I for one, don't feel comfortable in
>joining in unless I have listened for some period of time. Using trick
>coding systems is great, but it would be nice if there were one good
>standard defined that would cover both areas. Some state of the art
>tradeoffs are inevitable but it seems to me if we don't cover this
>ground at some point we are all going to be doing our own thing while
>the whole hobby sorta grinds down from the lack of general monitoring
>that leads to all sorts of interesting input (or output!) from people...
>
>Now,come on flames <grin>!
>
>de Jerry (wb6wpx)

Good suggestions - two comments:

1) Modern protocols matches the best modulation scheme with some form of
error control coding, often with time dispersed data interleaving. This is
purposely designed for to work with HF. These data streams need to be
synchronised and processed in its entirety - it is nearly impossible to
access and decode a partial data stream. Third party monitoring becomes
spotty, unreliable, and not much fun, I guess, because distant receivers
does not hear the exact same version of the transmitted signal. This is due
to the ionsphere playing tricks, also local interference playing its role.

2) There has been some slow speed BPSK activity on HF for some time - I can
only guess that it is somewhat more reliable than RTTY but not quite as good
as other modes that uses ECC. I have been told that G3PLX, PA0DCD and others
often have round-table QSO's using this mode. There now are M56002 and
TI320C50-based implementations on the air. There also is a very nice slow
speed BPSK package available for the Motorola EVM, the same that TAPR
offered a while ago. It would not take too much to start some grass-roots
activity. If anyone is interested in this, check out the TAPR DSP SIG's
software repository.

Johan

17. Options for clock extraction

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 11 Apr 1997 09:01:28 -0800
To: hfsig@tapr.org
Message-ID: <199704111557.IAA10286@PEAK.ORG>
Rendered body type: text/plain

Hi,

I wonder if anyone has a good idea to share on a reliable method for clock
extraction for an async data stream recovered from a 2-FSK signal.

This is for use on a data stream like for example 5-bit BAUDOT, 8-bit ASCII,
or 7-bit CCITT characters. The method must NOT rely on stop/start bits NOR

on bit edge transitions. This EXCLUDES ideas like zero crossing detectors,
interval/period gate timers, or one shots. Any random bit pattern is
possible, however it may be assumed that there would at least be one bit
transition per character frame.

The idea is to use the various signals from within the demodulator - any
suitable demodulator architecture is acceptable; passband, heterodyne, DPLL,
or whatever.

Any suggestions or ideas will be greatly appreciated.

Thanks much!

Johan

18. Re: [HFSIG:2068] Options for clock extraction

From: mcdermot@rockdal.aud.alcatel.com (Tom Mcdermott)
Date: Fri, 11 Apr 97 12:48:36 CDT
To: hfsig@tapr.org
Message-ID: <9704111748.AA01238@eagle.aud.alcatel.com>
Rendered body type: text/plain

>
> Hi,
>
> I wonder if anyone has a good idea to share on a reliable method for clock
> extraction for an async data stream recovered from a 2-FSK signal.
>
> This is for use on a data stream like for example 5-bit BAUDOT, 8-bit ASCII,
> or 7-bit CCITT characters. The method must NOT rely on stop/start bits NOR
> on bit edge transitions. This EXCLUDES ideas like zero crossing detectors,
> interval/period gate timers, or one shots. Any random bit pattern is
> possible, however it may be assumed that there would at least be one bit
> transition per character frame.
>
> The idea is to use the various signals from within the demodulator - any
> suitable demodulator architecture is acceptable; passband, heterodyne, DPLL,
> or whatever.
>
> Any suggestions or ideas will be greatly appreciated.
>
>
> Thanks much!
>
> Johan
>
>

        Johan: two questions about the conditions:

1. Are the stop bits an integer number of bit times long (i.e. 1x or 2x
   bit times)?

2. Is the source of the async data synchronous (i.e. does it always put the
   leading edge of the first event at a time that is an integer number of bits
   times removed from the last transmitted bit edge)?

If the answer is yes to both #1 and #2, then the data stream is fully
synchronous, and receive symbol timing can be recovered with classical
methods. This is detailed in my book in the section on symbol-timing (clock)
recovery. Basically, the methods are based on non-linear operations on the
received baseband data stream, and include rectification or squaring, with
squaring usually being the best, followed by filtering (filter or PLL). If the
data stream can go idle (no transitions) for a long time, then the design gets
very tricky.

If the answer to either question is no, then the data stream is not fully
synchronous (i.e.: the relative 'edge phase' of the data bits can change from
character to character). In this case, the design is more challenging, and
requires that a new synchronization epoch be detemined for each character.

One technique for doing this is to use a maximum-likelihood (ML) clock
position estimator. The idea is to store all of the digitized samples of the
received data stream, and to lay against that the several different possible
clock signals. If there are for example 32 samples per data bit, then there
are 32 possible clock phases for the character. Then, you derive an error
function of the databits against the clock signal. This is, of course,

the tricky part. Pick the clock phase that minimized the metric, and it is
the correct phase for that character (start, data, stop).

Some things to look for in an error metric:
* How close are the zero->one or one->zero crossings of the data to the correct
  time position (with respect to clock timing)?
* When is the time of maximum eye opening for each data bit, and how does that
  time correspond to the correct position w.r.t. clock timing?
* If you can identify the start and the stop bits, then you know ther value.
  How accurately do the received values reflect the known values (w.r.t.
  clock timing)?

Perhaps others can suggest a good technique for calcualting the error metric.

- Tom

Tom McDermott
mcdermot@aud.alcatel.com

19. Re: [HFSIG:2068] Options for clock extraction

From: Bill Carver <bcarver@magiclink.com>
Date: Fri, 11 Apr 1997 13:13:00 -0500 (CDT)
To: hfsig@tapr.org, hfsig@tapr.org
Message-ID: <199704111813.NAA03284@tapr.org>
Rendered body type: text/plain

As long as the character length is an integral number of bits you can just
make a PLL and phase lock a local clock to the incoming bit stream. I'm not
a guru like many of the guys on this SIG, but I phase locked a 45 Hz clock
to incoming transitions thirty years ago and ran point-to-point RTTY using
7.0 unit Western Union code. WA6JYJ and I ran a 200 mile path with
milliwatts with incredible reliability for the time.

At that time few were using WU machines, insted 7.42 unit code which it
would NOT lock to. I'm sure a similar thing can be done today and cope with
7.42 code and produce essentially the same result.

The question/comment will no doubt arise "WHY do you want to do this, we
have such advanced protocols this is obsolete". So lemme recognize that
question and ask in return:

WHY do YOU want high speed point-to-point communications? I'm not running a
traffic net. I want to sit on 10,136 and accurately monitor chatter among
10-15 people....some of them are temperature compensating boat anchor R390A
PTOs, others are talking about phase-linear crystal filters, and one guy has
a receiver mixer with a +56 dBm third order interecept and is trying to make
a local oscillator with low phase noise.

Is this asking for a high-accuracy FEC mode? If so, so be it. ARQ protocols
are only good for two people.

Regards - Bill - W7AAZ

20. Re: [HFSIG:2069] Re: Options for clock extraction

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 11 Apr 1997 14:37:38 -0800
To: hfsig@tapr.org
Message-ID: <199704112133.OAA27698@PEAK.ORG>
Rendered body type: text/plain

Hi Tom,

>
>   Johan: two questions about the conditions:
>
>1. Are the stop bits an integer number of bit times long (i.e. 1x or 2x
>   bit times)?

For this general discussion, lets not think of these as stop/start bits,
just that these are like any of the other bits and of the same duration.

>
>2. Is the source of the async data synchronous (i.e. does it always put the
>   leading edge of the first event at a time that is an integer number of bits

>   times removed from the last transmitted bit edge)?

With above definition then, lets assume that data is synchronous mode (on gaps inbetween characters), however there are no actual sync character(s) (such as HDLC flags) to derive epoch sync. One must derive bit syncs from the data itself.

Here is an example that may help illustrate what I mean:

```
 1   2   3  4    5   6   7   8    ....... bit numbers
   +---+       +---+       +---+
---+   +---+---+   +---+---+   +  ....... recovered data

\/\/\/\/\/\/\/\/\/\/\/\/\/\/\/\/\ ....... clock at 2x bit rate
   |   |   |   |   |   |   |   |
   a   b   c   d   e   f   g   h  ...... bit transitions
```

Shown above is an 8-bit data word (01001001). A perfectly synced, 2x data clock is shown below the bit stream. Note that at each bit transition shown, the phase of the 2x data clock is zero, i.e, one can assume that the bit transitions corresponds to the 2x data clock phase being zero.

The purpose of this exersize is to show the ability to track HF multipath affects on clock/bit recovery. I have the algorithm working so-so, but it needs reasonably good S/N. Ultimately I would like not to use bit edge extraction as that is the weakness in my opinion.

>
>One technique for doing this is to use a maximum-likelihood (ML) clock
>position estimator. The idea is to store all of the digitized samples of the
>received data stream, and to lay against that the several different possible
>clock signals. If there are for example 32 samples per data bit, then there
>are 32 possible clock phases for the character. Then, you derive an error
>function of the databits against the clock signal. This is, of course,
>the tricky part. Pick the clock phase that minimized the metric, and it is
>the correct phase for that character (start, data, stop).

This has may have some possibilities.

I was hoping that someone would point me to was some clever idea like you mentioned using a non-linear operation on the raw data stream that would yield a usable clock, however, at some raised frequency that one could use to lock a DPLL - like what is used to extract the "fantom" carrier from supressed-carrier waveform. Unfortuantely, I still am looking.

Thanks, 73

Johan

21. Re: [HFSIG:2068] Options for clock extraction

From: Lyle Johnson <lylej@azstarnet.com>
Date: Fri, 11 Apr 1997 18:13:10 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19970411175441.00724c98@pop.azstarnet.com>
Rendered body type: text/plain

Johan,

Years ago Paul Newland described his "Z-AMTOR" decoder (early 80s). To recover the clock initially he used a technique wherein he clocked in a number of bits worth of data with slightly offset clocks in a number of buffers. Hhhmm, that reads poorly.

Assume you have ten buffers and an assumed bit rate of 100 bit/sec (10 mSec/bit). You clock in the data with a 1 mSec clock, but only every tenth sample goes into a given buffer. When you are done, you have 10 copies of the data stream, but they are staggered. You then look at the buffers and find the one that matches the pattern you are seeking, or look for transitrions, etc., You then use that clock's timing (might be the 4th millisecond sample, for instance) and use that clock until you detect something is drifting.

With AMTOR this may work better than for some othe rmodes because AMTRO requires the bit clock to be very precise, but the method may be suitable for what you are trying to do?

Just a thought,

Cheers,

Lyle

22. Re: [HFSIG:2072] Re: Options for clock extraction

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 11 Apr 1997 22:42:43 -0800
To: hfsig@tapr.org
Message-ID: <199704120538.WAA24447@PEAK.ORG>
Rendered body type: text/plain

Hi Lyle,

>Johan,
>
>Years ago Paul Newland described his "Z-AMTOR" decoder (early 80s). To
>recover the clock initially he used a technique wherein he clocked in a
>number of bits worth of data with slightly offset clocks in a number of
>buffers. Hhhmm, that reads poorly.
>
>Assume you have ten buffers and an assumed bit rate of 100 bit/sec (10
>mSec/bit). You clock in the data with a 1 mSec clock, but only every tenth
>sample goes into a given buffer. When you are done, you have 10 copies of
>the data stream, but they are staggered. You then look at the buffers and
>find the one that matches the pattern you are seeking, or look for
>transitrions, etc., You then use that clock's timing (might be the 4th
>millisecond sample, for instance) and use that clock until you detect
>something is drifting.
>
>With AMTOR this may work better than for some othe rmodes because AMTRO
>requires the bit clock to be very precise, but the method may be suitable
>for what you are trying to do?
>
>Just a thought,
>
>Cheers,
>
>Lyle
>
>

Right, this is how we do AMTOR epoch sync. The next, and very formiddable task, is to track the other fellow's bit clock during the duration of the QSO. Besides multipath effects and noise, edges are not quite reliable, besides the other side is only required to maintain a 50 ppm clock, which is not very accurate.

My question thus relates as to how we do this kind bit tracking without relying so much on hard bit transitions? Traditionally this is done using a type of early-gate/late-gate synchronizer that works off thresholded data. G3PLX described this way back in the 1980's in his RADCOM articles when he first introduced AMTOR. Paul's Z-Amtor, I think also used a similar method, and I suspect most HF TNC's uses something along these lines. This is one reason that prevents AMTOR from working under very poor S/N conditions.

Johan

23. Re: [HFSIG:2073] Re: Options for clock extraction

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Sat, 12 Apr 1997 13:00:13 -0600 (MDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.95.970412124224.24778A-100000@acca>
Rendered body type: text/plain

Hi Johan,

I've been reading the recent thread with some interest and thought I'd make a few comments. My message kind of goes along with what Tom already mentioned I suppose.

>From your first message, I believe you are trying to extrace the bit rate clock to determine when to sample the signal and make a decision. I have been doing some of this recently for a BFSK signal and I implemented a nonlinearity followed by a DPLL approach Tom mentioned. After the incoming signal is bandpass filtered by each of the two FSK tone filters, and the filter outputs subtracted, I then take the absolute value of the difference. Of course on a DSP is this usually easy (abs a). The resulting PSD shows a large low frequency lobe that rolls off to a null at about 2x the bit rate, then a much smaller lobe. Anyhow, right at the bit rate is a

spike that is about 15-20 db above the lobe. I am planning to add a DPLL
to lock to this frequency component and act as a tight bandpass filter.
When the PLL rolls over from 2*pi back to 0 radians, I plan sample the
signal and make a decision. Of course I'm sure I have to figure out
exactly at what phase of the DPLL I need to sample at - I doubt it will be
right at 2*pi.

I was looking around for some notes on the HF channel model that you and
Tom spoke on in Seattle, but I couldn't find anything. I was going to see
how this worked with some multipath on the channel. I ran the test with an
image of the signal being delayed at 1mS at 50% amplitude and summed with
the original. The PSD did change somewhat, but the spike at the bit rate
wasonly about 13-14 db above the signal at that point. The eye also went
to crap, but that was expected! (time for that equalizer!)

I'm sure Tom may have some more comments. I ran this simulation on Matlab
on 1600 random FSK modulated bits(Bell 102 tones I think the spec is).

73
Tim

24. Re: [HFSIG:2074] Re: Options for clock extraction

From: forrerj@peak.org (Johan Forrer)
Date: Sat, 12 Apr 1997 14:37:20 -0800
To: hfsig@tapr.org
Message-ID: <199704122132.OAA19532@PEAK.ORG>
Rendered body type: text/plain

Tim,

>>From your first message, I believe you are trying to extrace the bit rate
>clock to determine when to sample the signal and make a decision.

Exactly.

I have had some other interesting e-mail asking why one needs this; Just
recon that when multipath affects results in say 1ms random delay per bit
(not typical, but not unusual for HF), an 8 bit, 10 ms/bit word can
theoretically vary in length between 8*9 = 72 ms and 8*11 = 88 ms (expected
= 80ms). These extremes is a range of 16ms! I think you can appreciate that
it becomes a challenge to rely on a start bit to tell you where to find each
bit. OK? Ideally one needs something that adapts on the fly to the
variations in path delay so that bit locations are adaptive and not
relative to some long-term fixed average. This concept is receiving some
attention in modern high speed HF modem research today. Typically this is
achieved through time/frequency diversity, or adaptive equalizers.

Hope this makes a bit more sense?

>I have
>been doing some of this recently for a BFSK signal and I implemented a
>nonlinearity followed by a DPLL approach Tom mentioned. After the incoming
>signal is bandpass filtered by each of the two FSK tone filters, and the
>filter outputs subtracted, I then take the absolute value of the
>difference. Of course on a DSP is this usually easy (abs a). The resulting
>PSD shows a large low frequency lobe that rolls off to a null at about 2x
>the bit rate, then a much smaller lobe. Anyhow, right at the bit rate is a
>spike that is about 15-20 db above the lobe. I am planning to add a DPLL
>to lock to this frequency component and act as a tight bandpass filter.
>When the PLL rolls over from 2*pi back to 0 radians, I plan sample the
>signal and make a decision. Of course I'm sure I have to figure out
>exactly at what phase of the DPLL I need to sample at - I doubt it will be
>right at 2*pi.

I think one can intuitively prove that you are right: If you use narrow
bandpass filter designed for twice the data rate, all it needs is data
pulses from either mark or space to make it ring it at its natural
resonating frequency (I leave it for the reader's to think about why a
bandpass filter at the data rate won't work for this purpose :)

>
>I was looking around for some notes on the HF channel model that you and
>Tom spoke on in Seattle, but I couldn't find anything. I was going to see
>how this worked with some multipath on the channel. I ran the test with an

>image of the signal being delayed at 1mS at 50% amplitude and summed with
>the original. The PSD did change somewhat, but the spike at the bit rate
>wasonly about 13-14 db above the signal at that point. The eye also went
>to crap, but that was expected! (time for that equalizer!)

Have a look at Eric Johnson's web pages for his PSK and m-FSK simulation
programs - you may get some good ideas on how he incorporates the Watterson
model into the simulation.

Thanks for the feedback. 73

Johan

25. Re: [HFSIG:2075] Re: Options for clock extraction

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Sat, 12 Apr 1997 23:17:27 -0600 (MDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.95.970412225401.8397B-100000@acca>
Rendered body type: text/plain

> I have had some other interesting e-mail asking why one needs this; Just
> recon that when multipath affects results in say 1ms random delay per bit
> (not typical, but not unusual for HF), an 8 bit, 10 ms/bit word can
> theoretically vary in length between 8*9 = 72 ms and 8*11 = 88 ms (expected
> = 80ms). These extremes is a range of 16ms! I think you can appreciate that
> it becomes a challenge to rely on a start bit to tell you where to find each
> bit. OK? Ideally one needs something that adapts on the fly to the
> variations in path delay so that bit locations are adaptive and not
> relative to some long-term fixed average. This concept is receiving some
> attention in modern high speed HF modem research today. Typically this is
> achieved through time/frequency diversity, or adaptive equalizers.
>
> Hope this makes a bit more sense?

Oh, yes, at least I believe I understand what your goal is. Sounds like
you just need a classical bit timing recovery and decision feedback
equalizer. I think the main concern would be setting up the equalizer so
as to adapt to the multipath changes on HF. From what I remember from the
session in Seattle, it doesn't seem to be changing all that fast. Tom, is
there any reason why an equalizer can't be used to adapt to the changes in
the HF channel?

Also, I've seen several emails about locating start bits, etc., for
timing. This seems to me to be up to the next layer in the system. I am
only thinking of the physical layer, treating all bits the same and
assumming the symbols are random with equal probability. Perhaps I missed
some earlier emails before this thread caught my attention?

> I think one can intuitively prove that you are right: If you use narrow
> bandpass filter designed for twice the data rate, all it needs is data
> pulses from either mark or space to make it ring it at its natural
> resonating frequency (I leave it for the reader's to think about why a
> bandpass filter at the data rate won't work for this purpose :)

Right, and a DPLL with a natural frequency at the bit rate will act as the
narrow bandpass filter. I'm interested in why 2x the data rate. For the
FSK stuff I'm working on, I've been seeing the spikes at the data rate,
and I believe a null at 2x the data rate.

I think some nonlinear clock recovery methods for other modulation schemes
do end up at twice the clock rate. (PSK where you square the signal or
raise it to the 4th?)

All my books tend to be at the office right now - and I'm not dedicated
enough yet to go to work on the weekends to get them! ;-)

> Have a look at Eric Johnson's web pages for his PSK and m-FSK simulation
> programs - you may get some good ideas on how he incorporates the Watterson
> model into the simulation.

Good idea.

Thanks for getting my interests sparked back up! Been too windy here in
Austin lately to do much ballooning, so other hobbies have been coming
back to me...

Tim

26. Re: [HFSIG:2076] Re: Options for clock extraction

From: Keith McElveen <jkm@mclean.sparta.com>
Date: Tue, 15 Apr 1997 10:45:49 -0400
To: hfsig@tapr.org
Message-ID: <3353949D.4D79@mclean.sparta.com>
Rendered body type: text/plain

> > I have had some other interesting e-mail asking why one needs this; Just
> > recon that when multipath affects results in say 1ms random delay per bit
> > (not typical, but not unusual for HF), an 8 bit, 10 ms/bit word can
> > theoretically vary in length between 8*9 = 72 ms and 8*11 = 88 ms (expected
> > = 80ms). These extremes is a range of 16ms! I think you can appreciate that
> > it becomes a challenge to rely on a start bit to tell you where to find each
> > bit. OK? Ideally one needs something that adapts on the fly to the
> > variations in path delay so that bit locations are adaptive and not
> > relative to some long-term fixed average. This concept is receiving some
> > attention in modern high speed HF modem research today. Typically this is
> > achieved through time/frequency diversity, or adaptive equalizers.
> >
> > Hope this makes a bit more sense?
>
> Oh, yes, at least I believe I understand what your goal is. Sounds like
> you just need a classical bit timing recovery and decision feedback
> equalizer. I think the main concern would be setting up the equalizer so
> as to adapt to the multipath changes on HF. From what I remember from the
> session in Seattle, it doesn't seem to be changing all that fast. Tom, is
> there any reason why an equalizer can't be used to adapt to the changes in
> the HF channel?.................

This is a very interesting thread, but it poses a problem that I have
been
trying to solve to no avail - what if you are not sending bits, but an
analog
voice signal instead? (I know that this is the SS group but you guys
seem to know what
is going on with multipath problems and equalizers, so I would
appreciate the
help immensely!). Can you adapt an equalizer to recombine the
multipaths
instead of just throwing out the late multipath energy with an LMS
adaptive
predictive deconvolutional filter? thanks for any help/insight that you
can provide!

27. Major progress with the NEWQPSK modem

From: Pawel Jalocha <JALOCHA@vxcern.cern.ch>
Date: Thu, 17 Apr 1997 15:58:33 +0200 (CET-DST)
To: hfsig@tapr.org
Message-ID: <970417155833.2066ff25@vxcern.cern.ch>
Rendered body type: text/plain

Hi all,

I am getting very good news recently from Timo OH6KK in Finland.
There they are testing the NEWQPSK modem (which was lying around useless :-)
for long time) with the aim to create links across Finland which is
not a densely populated country and the demand for fast HF links is strong.

Although the first test were not very encouraging the perspectives look
very good now. With certain precautions and after correcting the software
bugs the modem proofs to work like a normal packet even in the poorer
HF conditions on 80 meter.

The tests were done between Espoo and Konvola (100 miles or more)
on 80 meters in conditions ranging from very good (signals S9+20)
to very poor (S0). The top number I heard last weekend was 70 bytes/sec
on a 14 kB file and that what still with the software bug that the
Doppler drift tracker was correcting the wrong way which prevented
from doing more than 1 frame per transmition. The bug is found
and corrected now (one little minus sign somewhere in the code...)
and I expect better numbers to come through.

It has been observed that the modem does transfer data despite of
an in-band CW interference and SSB QSO's going on (with the strongest
FEC mode selected).
The other interresting observation was that although Timo and Joni
both use FT757, (same type, same year), and both TRX seem to perfom
normal on SSB, but the Joni's TX works real bad for the multi-tone signals.

This was determined when the third station: Villi, joined the tests
(he uses an ICOM 706).

People involved in these tests are:

Timo   OH6KK   sivula@ncsur01es.ntc.nokia.com
Joni   OH2NJR njr@clinet.fi
Villi   OH5LEP oulu@computec.fi
Hessu   OH7LZB hessu@pspt.fi or oh7lzb@zone.pspt.fi
Tomi   OH6LZX? tomi.manninen@ntc.nokia.com

The progress made by them is very encouraging for me
- you can see a light in the tunnel.
If anyone else wishes to join the tests please do so.
You need a PC (or KISS-doing packet radio software), the DSPCARD4
or EVM56K and a shortwave tranceiver with antenna and the ussuall
radio stuff.

Pawel

28. Re: [HFSIG:2078] Major progress with the NEWQPSK modem

From: Paul Mu <pmu@smarttalk.com>
Date: Thu, 17 Apr 1997 12:40:02 -0400
To: hfsig@tapr.org
Message-ID: <33565262.418B@smarttalk.com>
Rendered body type: text/plain

Hi,
  I would like to attend, but I am really a beginner in this area. I
have several PC. Could you tell me how can I get the rest of them such
as DSPCARD4, I have TM300 serials DSP, but where is DSP code ?

Thanks.
pmu@smarttalk.com

Pawel Jalocha wrote:
>
> Hi all,
>
> I am getting very good news recently from Timo OH6KK in Finland.
> There they are testing the NEWQPSK modem (which was lying around useless :-)
> for long time) with the aim to create links across Finland which is
> not a densely populated country and the demand for fast HF links is strong.
>
> Although the first test were not very encouraging the perspectives look
> very good now. With certain precautions and after correcting the software
> bugs the modem proofs to work like a normal packet even in the poorer
> HF conditions on 80 meter.
>
> The tests were done between Espoo and Konvola (100 miles or more)
> on 80 meters in conditions ranging from very good (signals S9+20)
> to very poor (S0). The top number I heard last weekend was 70 bytes/sec
> on a 14 kB file and that what still with the software bug that the
> Doppler drift tracker was correcting the wrong way which prevented
> from doing more than 1 frame per transmition. The bug is found
> and corrected now (one little minus sign somewhere in the code...)
> and I expect better numbers to come through.
>
> It has been observed that the modem does transfer data despite of
> an in-band CW interference and SSB QSO's going on (with the strongest
> FEC mode selected).
> The other interresting observation was that although Timo and Joni
> both use FT757, (same type, same year), and both TRX seem to perfom
> normal on SSB, but the Joni's TX works real bad for the multi-tone signals.
> This was determined when the third station: Villi, joined the tests
> (he uses an ICOM 706).
>
> People involved in these tests are:
>
> Timo   OH6KK   sivula@ncsur01es.ntc.nokia.com
> Joni   OH2NJR njr@clinet.fi
> Villi   OH5LEP oulu@computec.fi
> Hessu   OH7LZB hessu@pspt.fi or oh7lzb@zone.pspt.fi
> Tomi   OH6LZX? tomi.manninen@ntc.nokia.com
>
> The progress made by them is very encouraging for me
> - you can see a light in the tunnel.
> If anyone else wishes to join the tests please do so.
> You need a PC (or KISS-doing packet radio software), the DSPCARD4

> or EVM56K and a shortwave tranceiver with antenna and the ussuall
> radio stuff.
>
> Pawel

29. RE: [HFSIG:2079] Re: Major progress with the NEWQPSK modem

From: jalocha@axdele.cern.ch
Date: Thu, 17 Apr 1997 23:31:59 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199704172131.XAA22786@dxmint.cern.ch>
Rendered body type: text/plain

> I would like to attend, but I am really a beginner in this area. I
>have several PC. Could you tell me how can I get the rest of them such
>as DSPCARD4, I have TM300 serials DSP, but where is DSP code ?

The quickest is to get the EVM56002 package from a Motorola office for $150.
This includes the assembler and code downloader/debugger
for a PC. Or if you like soldering you can order the DSPCARD4
as a kit from Finland. There was somebody on this list offering
a mounted DSPCARD4 recently ?
The radio interface of the DSPCARD4 is well prepared while for the EVM56K
you need to do some wireing if you want proper PTT and UP/DOWN receiver
tuning, otherwise the cards are compatible (EVM56K's DSP is faster).
The DSP code for the modem you will get from me.
The BIOS you can get from the ftp.tapr.org.

Pawel

30. Re: [HFSIG:2079] Re: Major progress with the NEWQPSK modem

From: George Cserenyi <zsolt@direct.ca>
Date: Fri, 18 Apr 1997 17:16:40 -0700 (PDT)
To: Paul Mu <pmu@smarttalk.com>
Cc: hfsig@tapr.org
Message-ID: <Pine.LNX.3.95.970417175416.22612A-100000@ubc.ve7ciz.ampr.org>
Rendered body type: text/plain

On Thu, 17 Apr 1997, Paul Mu wrote:
> Hi,
>   I would like to attend, but I am really a beginner in this area. I
[...]
Where and when the meeting will be held?
I may attend if it is not too far from Vancouver BC.
George
[...]

31. RE: [HFSIG:2081] Re: Major progress with the NEWQPSK modem

From: jalocha@axdele.cern.ch
Date: Mon, 21 Apr 1997 15:11:24 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199704211311.PAA19239@dxmint.cern.ch>
Rendered body type: text/plain

Sorry for those who tried to contact Timo,
I have mis-spelled his e-mail address in my posting. It should be:

sivula@ncsnrs01es.ntc.nokia.com

that is I missed the "s" before the "0".

Pawel

32. Re: Options for clock extraction

From: Mike Klopfer <klopfer@eagle.natinst.com>
Date: Mon, 21 Apr 1997 22:39:29 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199704220339.WAA01627@eagle.natinst.com>
Rendered body type: text/plain

forrerj@peak.org (Johan Forrer) writes:

>I have had some other interesting e-mail asking why one needs this; Just
>recon that when multipath affects results in say 1ms random delay per bit
>(not typical, but not unusual for HF), an 8 bit, 10 ms/bit word can
>theoretically vary in length between 8*9 = 72 ms and 8*11 = 88 ms (expected
>= 80ms). These extremes is a range of 16ms!

I thought a multipath effect was due to for example a direct signal

and a reflected signal. Assuming that the source and reflector aren't
moving very fast relative to the receiver then the time for an 8 bit word
in worst
case would be when the first bit was from the direct path and the 8th
bit was from the reflector e.g 81ms for your example. I can't see how
you would get a cumulative error unless the clocks were drifting 1ms/10ms
or something was moving very fast.

Assuming there isn't such a drift problem I would guess that you could
find an equalizing filter before synchronizing. This could be done by
having the source transmit a known training sequence at periodic intervals
with enough silence in between to let any reflections die out. By sampling
at at least twice? the bit period you should be able to estimate the channel
impulse response.

Once an initial equalizer is obtained bit synchronization should be easy and
a DFE can be used to track the changes in channel response.

Of course there are also blind (no training sequence needed) methods of
finding the channel response but I don't know if they would be as simple
or reliable.

I would be interested in knowing any problems with this proposed solution.

mike

33. Re: [HFSIG:2083] Re: Options for clock extraction

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Fri, 25 Apr 1997 05:31:35 -0400
To: hfsig@tapr.org
Message-ID: <336079F7.7174@worldnet.att.net>
Rendered body type: text/plain

Mike Klopfer wrote:
>
>
> I thought a multipath effect was due to for example a direct signal
> and a reflected signal. Assuming that the source and reflector aren't
> moving very fast relative to the receiver then the time for an 8 bit word
> in worst
> mike

Mike:

The five or six layers in the ionosphere that are returning the five or
six different returns to you from the other station is moving and at
times quite rapidly with respect to BOTH observers. There is real
and observable doppler on HF signals.

Bob
--

****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

34. Re: [HFSIG:2084] Re: Options for clock extraction

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 28 Apr 1997 07:46:51 -0800
To: hfsig@tapr.org
Message-ID: <199704281439.HAA20466@PEAK.ORG>
Rendered body type: text/plain

Mike,

>Mike Klopfer wrote:
>>
>>
>> I thought a multipath effect was due to for example a direct signal

>> and a reflected signal. Assuming that the source and reflector aren't
>> moving very fast relative to the receiver then the time for an 8 bit word
>> in worst
>> mike
>
>Mike:
>
>The five or six layers in the ionosphere that are returning the five or
>six different returns to you from the other station is moving and at
>times quite rapidly with respect to BOTH observers. There is real
>and observable doppler on HF signals.
>
>Bob

Bob's note illustrates that this is a difficult situation; while recently
working on a new modem, I used a scope/logic state analyser to monitor
extracted bit edges from signals in the 6/8 mHz band. I was truly amazed at
the extent of this variability in timing at times when multipath was this
destructive. The use of some form of predictor/corrector mechanism such as
you indicated, does have its merits and is the topic of some on-going
reseach on HF modem development.

Actually, when dealing with good S/N situations, fairly simple solutions are
possible. However, things gets a little more challenging when S/N is not
ideal. What I was hinting at was that schemes that attempt to do data bit
extraction very late in the demodulator chain, using say for example, things
like simple diffentiators, are disadvantaged because a lot of good and
useful information that was present at passband/baseband, was disregarded.

73's

Johan

May 1997

Original source file: hfsig.9705
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9705

1. New DSP modems available

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 06 May 1997 08:03:13 -0800
To: dsp@tapr.org, hfsig@tapr.org
Message-ID: <199705061455.HAA00995@PEAK.ORG>
Rendered body type: text/plain

Hi folks,

Exciting new modems!

With Pawel's permission, I have uploaded his latest contributions:

NEWQPSK.ASM ----- Latest version of the 15-tone parallel 3000+ bps HF modem
FSK1200A.ASM ---- Heterodyne-based 1200 bps VHF Packet modem (very nicely done.)

As usual, these contain source code with a wealth of good ideas. It is up to
you to make your own modifications as needed, assemble the files, and run
them. If you use a standard EVM and BIOS, the code will run as is.

As soon as Tim has moved the files, please look for the package "MODEMS.ZIP"
in the "recent_uploads" file area of the TAPR DSP area.

Thanks Pawel!

Enjoy, 73

Johan

---

2. Re: [HFSIG:2086] New DSP modems available

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Tue, 6 May 1997 12:53:13 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.3.95.970506125056.19120G-100000@arctic>
Rendered body type: text/plain

I assume everyone on this sig is aware of the 100 quantity surplus 39 tone
Harris modems that are availabel for $100? I got some and they are brand
new in the box. I would think that although the 39 tone modems are 15
year old technology, that having 100 of them in the amateur community
would be a good basis for an amateur standard for some of the software DSP
versions you are testing. Also having a piece of working hardware aginast
whchi to test your DSP software should be of some value...

 On Tue, 6 May
1997, Johan Forrer wrote:

> Hi folks,
>
> Exciting new modems!
>
> With Pawel's permission, I have uploaded his latest contributions:
>
> NEWQPSK.ASM ----- Latest version of the 15-tone parallel 3000+ bps HF modem
> FSK1200A.ASM ---- Heterodyne-based 1200 bps VHF Packet modem (very nicely done.)
>
> As usual, these contain source code with a wealth of good ideas. It is up to
> you to make your own modifications as needed, assemble the files, and run
> them. If you use a standard EVM and BIOS, the code will run as is.
>
> As soon as Tim has moved the files, please look for the package "MODEMS.ZIP"
> in the "recent_uploads" file area of the TAPR DSP area.
>
> Thanks Pawel!
>
> Enjoy, 73
>
> Johan
>
>

APRSdos REPLY/COMMENT:

```
Reply mail addr:   wb4apr@amsat.org
US mail address:   115 old Farm Ct, Glen Burnie, MD 21060
See  APRS  LIVE:   http://web.usna.navy.mil/~bruninga/aprs.html
GPS on ANY radio:  http://www.tapr.org/tapr/html/mic-e.html
```

3. Re: [HFSIG:2086] New DSP modems available

```
From: Tim <tim@acca.nmsu.edu>
Date: Tue, 6 May 1997 13:19:07 -0600 (MDT)
To: hfsig@tapr.org
Cc: dsp@tapr.org
Message-ID: <Pine.SUN.3.95.970506131812.12552A-100000@acca>
Rendered body type: text/plain
```

On Tue, 6 May 1997, Johan Forrer wrote:

> Hi folks,
>
> Exciting new modems!
>
> NEWQPSK.ASM ----- Latest version of the 15-tone parallel 3000+ bps HF modem
> FSK1200A.ASM ---- Heterodyne-based 1200 bps VHF Packet modem (very nicely done.)
>
> As soon as Tim has moved the files, please look for the package "MODEMS.ZIP"
> in the "recent_uploads" file area of the TAPR DSP area.

Done!!!

73
Tim
AA5DF

4. S/N vs BW

```
From: dona@tridelta.com (Don Adams)
Date: Wed, 14 May 1997 11:10:01 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199705141510.LAA06826@sparcy.tridelta.com>
Rendered body type: text/plain
```

After all the years of coding discussions here, I thought
this Sig may be able to help. And it's been pretty quiet.

My question concerns channel capacity vs bandwith.

In shannon's classic equation C=W*log(1+(P/N))
where C=channel cappacity
        W= Bandwidth
        P=signal power
        N=noise power
This would imply that the channel capacity goes up
with increasing bandwidth. However the noise also
goes up with bandwith N=W*N0

I quickly plotted this on a spread sheet, C vs W,
Making P=1, N0=1, C limited around W=10 or so.

So bandwidth increases have their limits.

My question is: is this analysis true?

My real question is:
I belong to the SETI league.
This question arose from the fundimental assumption
that the signal we're all looking for is a very narrow band
CW, on the order of 1hz. The problem is finding that 1hz
in around 9Ghz of bandwidth. I'm questioning this basic
assumption. This seems to be a difficult signal to find (hi hi)
to say the least. If I were an ET, I would try to send something
that might be easier to find. Possibly some sort of orthoganal signal
(wavelets) that would allow any bandwidth receiver to detect/demod
the signal. Is it possible to spread out a signal so it may be easier
to find and possibly have a lower s/n threshold for detection?

Any comments?

Don wa8qzz

5. Re: [HFSIG:2089] S/N vs BW

From: Mike Fahmie <MPFahmie@lbl.gov>
Date: Wed, 14 May 1997 16:01:46 -0700
To: hfsig@tapr.org
Message-ID: <199705142301.QAA19397@mh1.lbl.gov>
Rendered body type: text/plain

At 10:15 AM 5/14/97 -0500, you wrote:
>After all the years of coding discussions here, I thought
>this Sig may be able to help. And it's been pretty quiet.
>
>My question concerns channel capacity vs bandwith.
>
>In shannon's classic equation C=W*log(1+(P/N))
>where C=channel cappacity
>   W= Bandwidth
>   P=signal power
>   N=noise power
>This would imply that the channel capacity goes up
>with increasing bandwidth. However the noise also
>goes up with bandwith N=W*N0
>
>I quickly plotted this on a spread sheet, C vs W,
>Making P=1, N0=1, C limited around W=10 or so.
>
>So bandwidth increases have their limits.
>
>My question is: is this analysis true?

In your example, the noise density remains constant as the signal density
decreases with bandwidth. Of course C limits, the signal is sinking below
the noise. Try your spreadsheet again and allow the signal power density
(P/W) to remain constant too.

-Mike-

                Mike Fahmie   -   Advanced Light Source
                Lawrence Berkeley National Laboratory
                (510) 486-4030   -   MPFahmie@lbl.gov
                                  WA6ZTY @ N6EEG

6. Re: [HFSIG:2089] S/N vs BW

From: Jack Taylor <jtaylor@huachuca-emh8.army.mil>
Date: Thu, 15 May 97 8:01:41 MST (1501Z)
To: hfsig@tapr.org
Message-ID: <199705151519.KAA25690@tapr.org>
Rendered body type: text/plain

Regarding the SETI project it too was my understanding they are scanning
for a CW or perhaps even a pulsed signal. While this may be alright,
wouldn't an advanced civilization be more apt to be using more advanced
communications techniques? Spread Spectrum comes to mind. Would think
the chances of detecting that would be pretty slim.
SSorry for the off-topic post.
773 de Jack

7. Re: [HFSIG:2090] Re: S/N vs BW

From: dona@tridelta.com (Don Adams)
Date: Thu, 15 May 1997 11:30:26 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199705151530.LAA12748@sparcy.tridelta.com>
Rendered body type: text/plain

>
>In your example, the noise density remains constant as the signal density
>decreases with bandwidth. Of course C limits, the signal is sinking below
>the noise. Try your spreadsheet again and allow the signal power density
>(P/W) to remain constant too.
>
>-Mike-
>

If U were tracking the signal, say with a phased locked loop, then
the signal to noise would be fixed by the PLL bandwith.
But for a spread spectrum signal, such as a direct sequence (vs freq hopping),
wouldn't the bandwidth of the receiver have to be wide?

 Or another situation
might be wide band FM vs narrow band fm. There's the FM advantage
when the peak deviation is sqr(2) times the highest modulation freq then

the s/n is equivalent to ssb. So if U want better s/n then U want a wider
deviation.
Isn't this an increase in channel capacity?

Back to PLL,would it be the different would be the lockup bandwidth vs the
tracking bandwidth.
I can see the PLL loop bandwith being multiplied by the number of
frequencies it can track.

???? Don ???

8. Re: [HFSIG:2089] S/N vs BW

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Thu, 15 May 1997 20:18:38 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199705151718.UAA06049@boojum.hut.fi>
Rendered body type: text/plain

This is really off-topic, I apologize...

> My real question is:
> I belong to the SETI league.
...
> to find and possibly have a lower s/n threshold for detection?
>
> Any comments?

Yes, from the point of view of physics. The question is: how small can we
measure? In the physics tendency seems to be to use high Q value antennas
(resonators) to detect the impossible. One example of this is the gravity
wave antenna. One of the latest proposals is a mechanical resonator 30 tons
(2.6 m diameter) of aluminium cooled to 0.01 K. The good point is that
the antenna can be cooled to so low temperature. For the 9 GHz radio
the antenna temperature is about 1 K. It is determined by the cosmic
background radiation. So my point is that wider bandwidth means less
sensitive antenna, if I understand everything correctly. For optimum CW
detection at 9 GHz you should have Q-value of 9 GHz / 500 Hz = 1.8 10^7.
This sounds like a superconducting cavity used in particle physics.
I hope the ET knows the CW code. Otherwise you may have to concentrate
to measure physical properties of the universe ;)

--
Jaakko Koivuniemi   ->   jaakko@boojum.hut.fi, tel 358-9-451 2973 (-2969 fax)
Low Temperature Laboratory, Helsinki University of Technology
Lab and vortices: http://boojum.hut.fi/ Me: http://boojum.hut.fi/%7Ejaakko/
On the air: OH7BF

9. Re: [HFSIG:2091] Re: S/N vs BW

From: dona@tridelta.com (Don Adams)
Date: Thu, 15 May 1997 14:06:44 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199705151806.OAA13832@sparcy.tridelta.com>
Rendered body type: text/plain

>Regarding the SETI project it too was my understanding they are scanning
>for a CW or perhaps even a pulsed signal. While this may be alright,
>wouldn't an advanced civilization be more apt to be using more advanced
>communications techniques? Spread Spectrum comes to mind. Would think
>the chances of detecting that would be pretty slim.
>SSorry for the off-topic post.
>773 de Jack
>
Hi Jack (oooooh) Hello Jack
Well that's the issue.....
If they don't want to be detected, I don't think we would find them.
However if they do want to be detected, is there a better signal
that could be sent. Something better than CW with a 1hz (or less) BW.
Something that's wider and easier to find. If it is a beacon, and if
radio is the fastest way to communicate, they may not be expecting a
reply. They may be broadcasting their life's history or something TBD....
So is there a better "symbol" that signifies "CQ" that we could lock on to
without knowing to much about them.
This isn't that much different from what we were doing when there were
discussions
about sync preambles. This is still a "thought experiment".... but if we
did think
of something better, maybe we could use it here. (This is the research part

at best)

over......Don wa8qzz

10. Re: [HFSIG:2089] S/N vs BW

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 15 May 1997 03:42:36 +0000 (UTC)
To: hfsig@tapr.org
Message-ID: <m0wRrR6-000RvpC@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>I quickly plotted this on a spread sheet, C vs W,
>Making P=1, N0=1, C limited around W=10 or so.

>So bandwidth increases have their limits.

>My question is: is this analysis true?

It certainly is. You've rediscovered the "Shannon Limit".

Let's rearrange the formulas a bit. Replace the signal power P with C*Eb, where C is the data rate in bits per second and Eb is the energy per bit (the product C*Eb is obviously still the signal power). And replace N with W * N0, where W is the bandwidth and N0 is the noise spectral density in watts/hz (again, the product is obviously equal to the total received noise power).

Now rearrange and solve for the ratio Eb/N0 as a function of W. Then take W to infinity. You'll get Eb/N0 = ln(2), which is equivalent to -1.6 dB. This is the famous Shannon Limit for the infinite bandwidth channel. Any realizable communication system must operate at or above a Eb/N0 of -1.6 dB. With finite channel bandwidth (say a few times the data rate) and the very latest implementable codes (called "turbo codes") the practical lower limit on Eb/N0 is a little less than +1 dB.

Phil

11. Re: [HFSIG:2089] S/N vs BW

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 15 May 1997 04:43:57 +0000 (UTC)
To: hfsig@tapr.org
Message-ID: <m0wRsOT-000RvuC@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>This question arose from the fundimental assumption
>that the signal we're all looking for is a very narrow band
>CW, on the order of 1hz. The problem is finding that 1hz
>in around 9Ghz of bandwidth. I'm questioning this basic
>assumption. This seems to be a difficult signal to find (hi hi)
>to say the least. If I were an ET, I would try to send something
>that might be easier to find. Possibly some sort of orthoganal signal
>(wavelets) that would allow any bandwidth receiver to detect/demod
>the signal. Is it possible to spread out a signal so it may be easier
>to find and possibly have a lower s/n threshold for detection?

This is a bit of a leap to a different subject (signal detection and estimation theory). You are right, if the purpose of SETI were to communicate at the highest possible data rate with the least RF power, the ET would do well to pick a strong code and an efficient modulation scheme.

The problem is, there are an astronomical number of highly efficient codes; it has been well known for a long time that any randomly chosen code, if it's long enough, is almost certainly quite efficient. So how would we know in advance which one is in use?

A *real* SETI "message" consists of only one "bit" meaning "this is clearly an artificial signal". The conventional wisdom has it that unmodulated narrowband carriers are the logical signal to use. There are no known natural mechanisms that can produce signals as narrow as the artifical methods we use (even natural masers are fairly broad), and the "search space", i.e., the spectrum, though large, is manageable.

Clearly, orthogonal sequences like walsh codes and semi-orthogonal sequences like PN codes would also indicate a signal of artificial origin, but the "search space" for such signals is so huge (especially when you combine them with the frequency space) that ET would probably not choose to use them.

Remember that sine waves of different frequencies also form orthogonal
signal sets. But among all possible orthogonal signal sets, Nature
seems to have a special place for sinusoids.

Whether the ETs out there (if any) who wish to advertise their
existence would follow the same reasoning as we have is, of course, a
bit of a leap of faith that has yet to pay off.

Bear in mind that you can easily look in multiple "places" (1Hz
frequency bins) at once with computational tools like the FFT. So in
principle it is really not a problem to search large sections of the
spectrum for signals as narrow as 1 Hz. There are many other practical
limits, such as spectral broadening in the interstellar medium (which
limits your integration time), and of course the big killer problem of
terrestrial QRM.

Phil

12. HAmPS Home Page

From: Marty Albert <marty@trucom.com>
Date: Fri, 16 May 1997 00:48:57 -0500
To: packet-pbbs@qsl.net, aprssig@tapr.org, bbssig@tapr.org, hfsig@tapr.org,
netsig@tapr.org, ss@tapr.org
Message-ID: <1.5.4.32.19970516054857.0069a7b8@mail.trucom.com>
Rendered body type: text/plain

This message is to announce the new Heartland AMateur Packet Society (HAmPS)
Home Page located at URL:

http://www.geocities.com/CapeCanaveral/Lab/7067

HAmPS is a group that facilitates packet radio network development with a
focus on operations in the Heartland of the United States.

Membership is FREE and open to all interested parties.

Please drop by the HAmPS Home Page for more information.

Take Care & 73

Marty Albert - marty@trucom.com
Amateur Radio: KC6UFM@KC6UFM.#SEMO.MO.USA.NOAM

Heartland Internet Services

****
Interested in Amateur Radio?
http://www.trucom.com/ppages/marty
****

13. Re: [HFSIG:2085] Re: Options for clock extraction

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Fri, 16 May 1997 05:32:29 -0400
To: hfsig@tapr.org
Message-ID: <337C29AD.C2E@worldnet.att.net>
Rendered body type: text/plain

Johan Forrer wrote:

> Bob's note illustrates that this is a difficult situation; while recently
> working on a new modem, I used a scope/logic state analyser to monitor
> extracted bit edges from signals in the 6/8 mHz band. I was truly amazed at
> the extent of this variability in timing at times when multipath was this
> destructive. The use of some form of predictor/corrector mechanism such as
> you indicated, does have its merits and is the topic of some on-going
> reseach on HF modem development.
>
> Actually, when dealing with good S/N situations, fairly simple solutions are
> possible. However, things gets a little more challenging when S/N is not
> ideal. What I was hinting at was that schemes that attempt to do data bit
> extraction very late in the demodulator chain, using say for example, things
> like simple diffentiators, are disadvantaged because a lot of good and
> useful information that was present at passband/baseband, was disregarded.
>
> 73's
>
> Johan

>

This is the primary reason for carrying along all of that "data-probe"
known
data information imbedded in the data in MIL-STD-188-110A and in Rohde &
Schwarz version of it as well as STANAG 4285. Since the most
distructive
multipath occurs usually on longer haul paths, the MIL-STD serial
(single
carrier modem) takes the data in LONG (4.8 second) blocks on these
really
long haul paths and permutes the order in which they are transmitted in
time. This spreads the burst errors out over 4.8 seconds (roughly)
providing
for diversity reception which aids the viterbi decoder to function even
in
the burst error environment. Even this is not enough. On the longer
haul
circuits, where the 4.8 second permuter (called an interleaver) is used
rather than the 0.6 second version, and where typically the bit rate is
1200 bps or less, 20 known symbols are transmitted for every 20 unknown
symbols. So after you have FEC'd your data, you interleave it (permute
the time order) and then you break the data up into 20 symbol chunks (
not 20 bits as the number of bits/symbol varies with the bit rate).
These 20 known symbols allow for recover of channel parameters in some
of
the worst multipath I have ever seen. Where the selective fading has
put
20-30 dB notches in multiple places in the spectrum and runs the baud
timing
all over the place. This is because it enables you to recover a matched
filter operating at N (where N is almost always 2) times per baud
operating
only over 60 symbol times: 20 known, 20 unknown, 20 known. That is,
you
"surround" the unknown data with known data. This enables the matched
filter to do all of the clock recovery for you. You sample its output
20
times and if the timing is not stable enough for a lousy 20 symbol times
(8 ms), you had no business trying to communicate any way. If you have
a
catastrophic loss of timing, in the 20 bauds, you will be likely except
in
the worst cases, to recover the timing on the next 20 unknown bauds, and
put the data in the right place in the interleaver. This 20 bauds that
get fouled up will be spread out and will be a big burst of errors. I
have
watched modems, including my own, go right over these bad patches of
data
with no problem, over 3000 mile paths at 1200 bps and over the pole at
150
bps. I do not know why we would want to reinvent the wheel. If we
emulated MIL-STD-188-110A, incorporating as it does FEC, time diversity,
etc., variable bit rates which can be made automatic, I have a hard time
understanding how we could do a lot better but I am willing to listen.
We would need to modify the standard for our purposes, since we would
almost surely want to easily incorporate ARQ, without having to send 4.8
seconds of data to do so. However, I caution you, there is no free
lunch on HF. You have to pay the penalties required to operate in
this medium.

Bob

--

****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

14. Re: [HFSIG:2096] Re: S/N vs BW

From: Glen Worstell <Glen_Worstell@notes.seagate.com>
Date: 16 May 97 13:54:39
To: hfsig <hfsig@tapr.org>
Message-ID: <9705161855.AA8241@notes.seagate.com>
Rendered body type: text/plain

>This question arose from the fundimental assumption
>that the signal we're all looking for is a very narrow band
>CW, on the order of 1hz.   ...

This is starting to get away from hfsig, but it is quite interesting.
I'd like to ask one question: Why do we assume that "they"
may be sending a signal to "us", when "we" are not sending
any signals to "them"?

Glen, KG0T

15. Re: Seti

From: k5cnf1@juno.com (Richard B Morrow)
Date: Fri, 16 May 1997 20:31:47 PST
To: hfsig@tapr.org
Message-ID: <19970516.205235.7607.0.k5cnf1@juno.com>
Rendered body type: text/plain

I don't know if anyone has heard of Project Serendip which explored much
lower frequencies than microwave, such as 425-435 MHz and found a bunch
of choerent sigs. As far as et sigs are concerned, why not operate on
the assumption that they will be no different than what we normally use,
such as traffic between science stations or outposts or whatever. It is
not going to be easy to find them, as the celestial mechanics are really
wild.

73, Richard K5CNF

16. Re: [HFSIG:2091] Re: S/N vs BW

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 16 May 1997 17:21:56 +0000 (UTC)
To: hfsig@tapr.org
Message-ID: <m0wSQhY-000RvhC@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>Regarding the SETI project it too was my understanding they are scanning
>for a CW or perhaps even a pulsed signal. While this may be alright,
>wouldn't an advanced civilization be more apt to be using more advanced
>communications techniques? Spread Spectrum comes to mind. Would think
>the chances of detecting that would be pretty slim.

An ET would probably not choose a spread spectrum signal if he were
deliberately trying to attract attention even if he knew that his
target "audience" understood spread spectrum. The problem is that the
"search space" is just too huge. Even unmodulated carriers represent
an enormous search space when you consider the size of the microwave
spectrum and the narrow channels (long integration times) required
to gather enough energy for detectability over interstellar distances.

Phil

17. Re: [HFSIG:2092] Re: S/N vs BW

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 16 May 1997 18:34:00 +0000 (UTC)
To: hfsig@tapr.org
Message-ID: <m0wSRpI-000RwzC@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>But for a spread spectrum signal, such as a direct sequence (vs freq hopping),
>wouldn't the bandwidth of the receiver have to be wide?

Before despreading, yes. But after direct sequence despreading the
signal bandwidth returns to its original narrow value. And if you're
trying to track a coherent carrier component within the despread
signal, a carrier tracking loop will strip off the modulation and
leave you with just the (reconstructed) carrier. You can make the
tracking loop for this carrier as narrow as you want, limited only by
the coherence time/doppler broadening of the channel.

> Or another situation
>might be wide band FM vs narrow band fm. There's the FM advantage
>when the peak deviation is sqr(2) times the highest modulation freq then

>the s/n is equivalent to ssb. So if U want better s/n then U want a wider
>deviation.
>Isn't this an increase in channel capacity?

If you widen the FM signal too much, the SNR drops below the FM
threshold and you're stuck. You can get some SNR back with a tracking
PLL demodulator, but this works only if the baseband signal bandwidth
isn't too wide (since the PLL needs to be able to follow the modulation).

>Back to PLL,would it be the different would be the lockup bandwidth vs the
>tracking bandwidth.
>I can see the PLL loop bandwith being multiplied by the number of
>frequencies it can track.

No, as I said above you track carrier in a direct sequence system
after despreading, so it's just the same as in a non-spread system.

The lockup/tracking bandwidth issue is relevant to initial acquisition, but not
to steady-state tracking and demodulation.

I point out that many real-world environments involve fast fading where coherent
carrier tracking isn't practical. These systems either use noncoherently demodulated
schemes (which don't require a receiver carrier phase reference) or send a "pilot"
reference along with the signal that can be used as the local carrier reference at
the receiver. Both approaches are used in IS-95 CDMA digital cellular: the first
(noncoherent) for the reverse link and the second for the forward link.

Phil

18. Re: [HFSIG:2101] Re: S/N vs BW

From: sams@jetlink.net (Sam Stanifer)
Date: Fri, 16 May 1997 22:36:13 +0800
To: hfsig@tapr.org
Message-ID: <v01530504afa21b9a0911@[206.72.64.152]>
Rendered body type: text/plain

>>Regarding the SETI project it too was my understanding they are scanning
>>for a CW or perhaps even a pulsed signal. While this may be alright,
>>wouldn't an advanced civilization be more apt to be using more advanced
>>communications techniques? Spread Spectrum comes to mind. Would think
>>the chances of detecting that would be pretty slim.
>
>An ET would probably not choose a spread spectrum signal if he were
>deliberately trying to attract attention even if he knew that his
>target "audience" understood spread spectrum. The problem is that the
>"search space" is just too huge. Even unmodulated carriers represent
>an enormous search space when you consider the size of the microwave
>spectrum and the narrow channels (long integration times) required
>to gather enough energy for detectability over interstellar distances.
>
If an ET realized the enormity of this problem, why would they bother? For
example, we listen, but do we transmit? No, because the resources required
far exceed the low probability of success. Listening is one thing, but
generating megawatts for a "beacon" is another.

Furthermore, a likely "target" will probably be a planet with similiar
distances from it's sun as we have from ours. Even if we knew precisely
where to point,the narrowest beamwidth we can generate would encompass the
entire solar system of the target planet at interstellar distances. What
are the chances that an ET could generate enough ERP to overcome solar
noise?

Not only that, but everyone knows they communicate with telepathy, so why
do they need electromagnetic communications at all?

(enjoy the ride)
73
Sam
KJ6BU

19. Re: S/N vs BW

From: k5cnf1@juno.com (Richard B Morrow)
Date: Sat, 17 May 1997 00:35:09 PST
To: hfsig@tapr.org
Message-ID: <19970517.003828.4487.0.k5cnf1@juno.com>
Rendered body type: text/plain

On Fri, 16 May 1997 17:08:18 -0500 (CDT) Glen Worstell
<Glen_Worstell@notes.seagate.com> writes:

>>This question arose from the fundimental assumption
>>that the signal we're all looking for is a very narrow band
>>CW, on the order of 1hz.   ...
>
>This is starting to get away from hfsig, but it is quite interesting.
>I'd like to ask one question: Why do we assume that "they"
>may be sending a signal to "us", when "we" are not sending
>any signals to "them"?
>
My neighbors portable phone is not sending any signals to me but I can
recieve them. Same thing with ET signals, IF we knew where to look and
when.

K5CNF

20. Re: [HFSIG:2095] Re: S/N vs BW

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Sun, 18 May 1997 13:15:15 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199705181015.NAA04813@boojum.hut.fi>
Rendered body type: text/plain

Let us reformulate the ETX problem: given discrete array of samples x[i]
how do we know that this represents a "modulated" signal? I am sure there
are number of radio sources in the galaxy where the Nature is modulating
the signal (neutron stars etc.). One obvious way is to look for correlations
between two samples x[i] and x[j]. There should be strong correlation, but
not too strong to allow some information to be passed. Next question is
how do we interpreter(i.e. demodulate) this language? Possible application:
intelligent demodulator, which will make CW/SSB/RTTY/AMTOR/PACTOR...
understandable automatically (down on the earth!).

--
Jaakko Koivuniemi   ->   jaakko@boojum.hut.fi, tel 358-9-451 2973 (-2969 fax)
Low Temperature Laboratory, Helsinki University of Technology
Lab and vortices: http://boojum.hut.fi/ Me: http://boojum.hut.fi/%7Ejaakko/
OH7BF

21. Re: [HFSIG:2105] Re: S/N vs BW

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sun, 18 May 1997 07:01:08 -0400
To: hfsig@tapr.org
Message-ID: <337EE174.56F5@worldnet.att.net>
Rendered body type: text/plain

Jaakko Koivuniemi wrote:
>
> Let us reformulate the ETX problem: given discrete array of samples x[i]
> how do we know that this represents a "modulated" signal? I am sure there
> Lab and vortices: http://boojum.hut.fi/ Me: http://boojum.hut.fi/%7Ejaakko/
> OH7BF

If I were an ET, I would send pulses. They can be so large in amplitude
that a blind bat with a coat hanger for an antenna can see them. They
require NO specialized knowledge/equipment such as spreading codes
and phase recovery in the spreading codes. They don't assume major
league coherence the way PSK or even straight carriers do for proper
gain by using long integration times. I think only pulses make sense.
When I was asked to do technology transfer to SETI folks, the thing
discussed with Sagan, Barney Oliver, Frank Drake, Dave Horowitz, etc.
was techniques for detection of these type signals but we offered to
them the optical spectrum analysis tools in order to enable them to
detect pulses: the R.F. modulates a crystal which is being illuminated
by a laser. The diffraction pattern is an amplitude spectrum. These
devices usually suffer from poor dynamic ranges with one major exception
and that is detecting high power pulses, especially those with decent
pulse repetition frequencies. Remember that you can run one watt into
a charging device such as a capacitor and then release it extremely
quickly and produce a huge power pulse. This is a slight exaggeration
of what is usually done but designed to get you to think about what
power is: energy consumed/time.

--
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,

```
*
* (H) 609-443-8963         * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061         * Buffalo (NE-III-120).  George Washington
*
* (W) 609-279-6240         * Council.  Brownie Troop 193
*
*************************************************************************
```

22. Re: ETBW

From: k5cnf1@juno.com (Richard B Morrow)
Date: Sun, 18 May 1997 20:08:25 PST
To: hfsig@tapr.org
Message-ID: <19970518.202354.4479.1.k5cnf1@juno.com>
Rendered body type: text/plain

Hey everyone,
why can't ET just be sending normal traffic like we do? Don't know what
frequencies they are on, but I will bet that the signals will have to
follow the same laws that ours do. Both emission wise and propagation
wise.

RM

23. Re: S/N vs BW

From: k5cnf1@juno.com (Richard B Morrow)
Date: Sun, 18 May 1997 20:22:16 PST
To: hfsig@tapr.org
Message-ID: <19970518.202354.4479.3.k5cnf1@juno.com>
Rendered body type: text/plain

On Sun, 18 May 1997 05:19:03 -0500 (CDT) Jaakko Koivuniemi
<jaakko@boojum.hut.fi> writes:
>
>Let us reformulate the ETX problem: given discrete array of samples
>x[i]
>how do we know that this represents a "modulated" signal? I am sure
>there
>are number of radio sources in the galaxy where the Nature is
>modulating
>the signal (neutron stars etc.). One obvious way is to look for
>correlations
>between two samples x[i] and x[j]. There should be strong correlation,
>but
>not too strong to allow some information to be passed. Next question
>is
>how do we interpreter(i.e. demodulate) this language? Possible
>application:
>intelligent demodulator, which will make CW/SSB/RTTY/AMTOR/PACTOR...
>understandable automatically (down on the earth!).

An intelligent demodulator is a good idea, who will be first?

K5CNF

24. Re: [HFSIG:2106] Re: S/N vs BW

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Mon, 19 May 1997 11:34:59 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199705190834.LAA14937@boojum.hut.fi>
Rendered body type: text/plain

> If I were an ET, I would send pulses. They can be so large in amplitude
...
> power is: energy consumed/time.

Maybe you are right. One example where the Nature is doing the "modulatio" is
telegraphic noise. There the system is fluctuating between two levels
in random way. This used to be a problem with the early transistors (I have
been told so). Now you may spend hours and hours to try to figure out
the message, which may sound like a funny Morse code. The oscillator
to produce RF signal does not need to be a lumped LC-circuit or cavity, it can
be also motion of mass, particles etc. Just to make more clear what I mean...

Jaakko/OH7BF

25. Re: [HFSIG:2106] Re: S/N vs BW

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 19 May 1997 23:55:45 +0000 (UTC)
To: hfsig@tapr.org
Message-ID: <m0wTcHJ-000RuZC@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>If I were an ET, I would send pulses. They can be so large in amplitude

There's only one problem: natural sources can generate highly regular
pulses. They're called pulsars. When they were first discovered,
they were thought to be a sign of intelligent life.

The reasoning behind listening for narrowband CW signals as signs of
ETs is that there are no known natural mechanisms for generating
extremely narrow coherent RF energy. And they're also an artifact of
many known modulation methods.

Phil

26. Re: [HFSIG:2110] Re: S/N vs BW

From: dona@tridelta.com (Don Adams)
Date: Wed, 21 May 1997 10:39:29 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199705211439.KAA06634@sparcy.tridelta.com>
Rendered body type: text/plain

>The reasoning behind listening for narrowband CW signals as signs of
>ETs is that there are no known natural mechanisms for generating
>extremely narrow coherent RF energy. And they're also an artifact of
>many known modulation methods.
>
>Phil

So basically, Phil, Ur saying that a CW signal or a signal with a lot of
carrier in it, is the best known way to indicate an un-naturally produced
signal......   I was questioning some of these basic assumptions for SETI.
I was looking, possibly, for some new math that might be able to be a
better,(faster stronger) way to flag an ET signal. In general Math can be
20- 50 years ahead of any practical applications. So I guess we're stuck
with all the doppler problems due to the complex relative motions of objects
in the universe. Somewhat like the multipath and unstable behavior of the
ionosphere.
 I was hoping for was some new radical (crazy) ideas. In general
I have found that really new ideas come from really crazy ones. Many times
when U scrape off the crazy part, that causes most people to discard it,
there is a nugget there that can be developed into something really new...
If U take the same old tract, U get evolutionary ideas, not revolutionary. I can
hardly, I mean not, think of any famous inventor or scientist, who was not
considered "crazy" or eccentric in his life time.

Many of the questions or comments on this thread have been considered
by the SETI community. There is ALOT of SETI stuff on the net.
The seti league is a good place to start. Especially if U want to get into
microwaves
with some socially redeeming purpose. (Although 1.4Ghz is barely microwaves
today)
Or have some use for that old satellite dish, now that they've scrambled
everything but NASA.
http://seti1.setileague.org/
The following article is relatively long (~20 pages) but has a reasonable
discussion
of the types of civilizations that might be out there and if so, what
evidence they
may be radiating.
http://www.setiquest.com/article1.htm

over and out......Don wa8qzz

27. Re: [HFSIG:2111] Re: S/N vs BW

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Thu, 22 May 1997 09:53:13 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199705220653.JAA10559@boojum.hut.fi>
Rendered body type: text/plain

> So basically, Phil, Ur saying that a CW signal or a signal with a lot of
> carrier in it, is the best known way to indicate an un-naturally produced

<deleted>
> hardly, I mean not, think of any famous inventor or scientist, who was not
> considered "crazy" or eccentric in his life time.

Many of the hams are already considered eccentric by others ;-). I think
you definitely need a powerful super/parallel computer to do the search
in the spectrum. You are looking for rare or weak radio events. It is
unlikely that just by tuning the VFO knob you will accidentally find such
a signal. It is not hard to guess that similar technology is already used
in reconnaissance. The possible new technology you are refering to will
take some time before it is in our hands. For the time being you can
enjoy the fb radio and antenna you have. Good luck!

Jaakko/OH7BF

28. Negative SNR, SS and FEC.

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Wed, 21 May 1997 21:05:53 +0200
To: hfsig@tapr.org
Message-ID: <s3840dd2.030@KIDD.CO.ZA>
Rendered body type: text/plain

Does someone have the time to explain the following in layman's
terms:(or perhaps in undergrad maths :-)

1. It is said that a spread spectrum system can work at negative SNR's.
How does this tie in with Shannons Limit of -1.6dB ?

2. What is the difference between spread spectrum process gain and
coding gain that one can obtain from say Viterbi FEC ?

Just wondering, the SETI thread has made me think about it.

Danie

29. RE: [HFSIG:2113] Negative SNR, SS and FEC.

From: "Borcher, David (NG-MS)" <DBORCHER@nogmail.gi.com>
Date: Thu, 22 May 1997 07:42:00 -0700 (PDT)
To: hfsig <hfsig@tapr.org>
Message-ID: <338458E4@gismtpgate.gi.com>
Rendered body type: text/plain

>1. It is said that a spread spectrum system can work at negative SNR's.
> How does this tie in with Shannons Limit of -1.6dB ?

The common misconception of SS "negative SNR" is in the
terminology - The spread signal when observed across its' occupied
bandwidth may be negative SNR but when the signal is de-spread
back to its' constituent data stream it is subject to the same SNR
constraints as any other signal of the same modulation type.

The SS process gain is pretty much the ratio of spread chips-per-bit over
the data rate in bps. Or approximated by the ratio of filter bandwidths
for spread vs de-spread to pass the signal without introducing further
distortions.

As an example, I worked on one of the firrst civilian applications of
direct-sequence SS - A VSAT satellite system - the data rate was
9600 BPS (BPSK) and the spreading was 256 chips-per-bit for a
2.4576 MHz RF modulation rate. The downlink signal could not be
seen above the noise floor of the LNA behind a 30" dish but the
despread 9.6 KB/s data generally had a 10 dB Eb/No and resulted
in a 1E-5 BER uncorrected - pretty much what the BER waterfall
curves would predict.

Dave K7NG

From: hfsig
Sent: Thursday, May 22, 1997 2:12 AM
To: hfsig
Subject: [HFSIG:2113] Negative SNR, SS and FEC.

Does someone have the time to explain the following in layman's
terms:(or perhaps in undergrad maths :-)

2. What is the difference between spread spectrum process gain and
coding gain that one can obtain from say Viterbi FEC ?

Just wondering, the SETI thread has made me think about it.

Danie

30. Re: [HFSIG:2110] HFSIG digest 320

From: w1gak@mindport.net
Date: Thu, 22 May 1997 12:06:24 -0400
To: hfsig@tapr.org
Message-ID: <199705221606.MAA05504@Eden.mindport.net>
Rendered body type: text/plain

three cheers for CW

31. Re: [SS:1458] [HFSIG:2113] Negative SNR, SS and FEC.

From: jra1854@tntech.edu (Jeffrey Austen)
Date: Thu, 22 May 1997 15:49:59 -0600
To: ss@tapr.org
Cc: hfsig@tapr.org, RLANIER@mailb.harris.com (RLANIER)
Message-ID: <v01540b00afaa69adf08d@[149.149.39.26]>
Rendered body type: text/plain

>Does someone have the time to explain the following in layman's
>terms:(or perhaps in undergrad maths :-)
>
>1. It is said that a spread spectrum system can work at negative SNR's.
>How does this tie in with Shannons Limit of -1.6dB ?

Both are correct. The confusion arises because they refer to two different measurements.

Shannon's limit states that reliable communication can be obtained with Eb/No >= -1.6 dB. Eb is the energy per data bit (before error correction coding -- coding may make the bit or symbol rate in the channel different than the actual data rate) and No is the noise power spectral density.

SNR is signal to noise ratio, that is signal power divided by noise power.

Eb/No is related to SNR by the data rate and the effective noise bandwidth. The following relates SNR to Eb/No. Ps is signal power and Pn is noise power. (The noise power spectral density is assumed to be constant.)

$$\frac{Ps}{Pn} = \frac{\text{Bit Rate}}{\text{Bandwidth}} \times \frac{Eb}{No}$$

In a spread-spectrum system the same power is spread out over a much larger bandwidth. From the above it can be seen that if the bandwidth is increased by a factor of one-hundred that the SNR will decrease by the same amount. Thus the SNR can be negative while Eb/No is positive.

>2. What is the difference between spread spectrum process gain and
>coding gain that one can obtain from say Viterbi FEC ?

Process gain and coding gain are different things. Process gain is a measure of the amount of "spreading" which is done to the signal. It is directly related to the interference rejection capabilities. If the channel is ideal (additive white Gaussian noise, no interference, no fading) process gain does absolutely nothing towards better error rates. However, in a real channel (with interference, fading, etc.) process gain does help mitigate some of the effects and therefore does help the error rate as compared to a non-spread system. Coding gain is obtained from adding redundancy to the message sent.

>Just wondering, the SETI thread has made me think about it.

Sounds interesting. Looking at how communication system have evolved over the past few decades makes me wonder how we expect to receive whatever type of modulation the ETIs decide to send!

Jeff, k9ja

P.S. I'm currently not on HFSIG and will not see followups to this message from that group.

Jeffrey Austen | Tennessee Technological University
jra1854@tntech.edu | Box 5004

+1 615 372 3485 | Cookeville Tennessee 38505 U.S.A.

32. FEC & White Noise

From: zs6awk@global.co.za (Danie Brynard)
Date: Sat, 24 May 1997 16:39:40 +0200
To: hfsig@tapr.org
Message-ID: <199705241431.QAA08482@lin01.global.co.za>
Rendered body type: text/plain

Are FEC (ie Viterbi) techniques generally only effective against white noise ?

Danie

++
Danie Brynard Email: zs6awk@global.co.za
PO Box 15133,Lyttelton,0140,South Africa
GRID LOC: KG44CE, Tel/Fax: (2712) 6640330
++

33. Re: [HFSIG:2111] Re: S/N vs BW

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 28 May 1997 22:16:28 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199705290516.WAA26027@servo.qualcomm.com>
Rendered body type: text/plain

Catching up on old email..

>So basically, Phil, Ur saying that a CW signal or a signal with a lot of
>carrier in it, is the best known way to indicate an un-naturally produced
>signal......

Correct. This has been the Conventional Wisdom in SETI for some time.

>I was looking, possibly, for some new math that might be able to be a
>better,(faster stronger) way to flag an ET signal. In general Math can be

Only I don't know of any. As I said earlier, the SETI "search space"
is already uncomfortably large even when we use the set of signals
(sinusoids) with the most efficient search algorithms known, namely
the FFT, which is complexity O(N log N).

This thread is getting rather far afield of HF modulation, so we should
probably draw it to a close.

Phil

34. Re: [HFSIG:2113] Negative SNR, SS and FEC.

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 29 May 1997 01:06:19 -0700 (PDT)
To: hfsig@tapr.org, ss@tapr.org
Message-ID: <199705290806.BAA26416@servo.qualcomm.com>
Rendered body type: text/plain

>1. It is said that a spread spectrum system can work at negative SNR's.
>How does this tie in with Shannons Limit of -1.6dB ?

The other responders have answered this well, so I'll just reiterate
that SNR and Eb/N0 are not the same thing; they're related by a factor
equal to the bit rate divided by the bandwidth. For a spread system,
the SNR in the full spread bandwidth is much lower than the Eb/N0.
No system, spread or not, can operate at a Eb/N0 < -1.6 dB.

>2. What is the difference between spread spectrum process gain and
>coding gain that one can obtain from say Viterbi FEC ?

Both process gain and coding gain can lower the required SNR in the RF
channel bandwidth, because both have the effect of spreading the
signal energy over a wider bandwidth. But only coding gain can lower
the required Eb/N0 (but never below -1.6 dB).

Phil

35. Announcement: Talks at SeaPac.

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 29 May 1997 08:46:43 -0800
To: hfsig@tapr.org, dsp@tapr.org

Message-ID: <199705291535.IAA26481@peak.org>
Rendered body type: text/plain

Hi folks,

Thought I would like to let you know of an interesting event for
those living in the Northwest USA:

SeaPac, the ARRL, N.W. Convention, puts on a number of talks each year at
the Seaside, Oregon, Convention. This year, there are two that may
be of interest to those following this SIG:

---
1) Sun 15th June, 9:15-10:15 AM. "Future Trends in HF Digital Communications."
   Part I: The Ionosphere; its effects on HF digital communications;
            HF channel simulation for testing and evaluation.
   Part II: Contemporary HF modem design; high speed parallel and
            serial tone modems; application of coding theory.
   Speaker: Johan Forrer, KC7WW
---
2) Sun 15th June, 10:45 AM. "DSP Basics."
   Speaker: C. Uyeda, KJ6HC.

There also are other interesting talks and workshops put on by the AMSAT
folks, topics
such as "HF Amplifier Design" by W7RQT, also Roy Lewallen on "Computer
Antenna Analysis"
(the latter two are on Saturday, 14th June.)

These talks are usually well-attended events, quite informal and stimulates
discussion.
If you are in the neighborhood, consider looking into this.

73,

--Johan

36. Re: [HFSIG:2117] FEC & White Noise

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 29 May 1997 23:07:06 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199705300607.XAA11006@servo.qualcomm.com>
Rendered body type: text/plain

>Are FEC (ie Viterbi) techniques generally only effective against white noise ?

No. With sufficient interleaving, FEC is even more effective against
fading and pulsed interference than against white noise.

In terms of the power required to attain a certain required bit error
rate, the gains from coding on a Rayleigh fading channel can be as
much as 30-40 dB.

Against strong pulsed interference the "gains" can be effectively
infinite, as it may be simply impossible to run enough power to
overcome the peak interference power. With proper coding and
interleaving, you don't have to.

Phil

June 1997

Original source file: hfsig.9706
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9706

1. unsubscribe

From: Guillaume <guillaum@worldonline.nl>
Date: Sun, 01 Jun 1997 14:19:28 +0200
To: hfsig@tapr.org
Message-ID: <3.0.1.32.19970601141928.006a3ba8@worldonline.nl>
Rendered body type: text/plain

[Empty message body]

2. unsubscribe

From: Ed Lyons <eel@ipass.net>
Date: Sun, 01 Jun 1997 15:16:26 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970601191626.00688bfc@ipass.net>
Rendered body type: text/plain

[Empty message body]

3. unsubscribe

From: "Robson Ribeiro da Silva" <aresta-eng@originet.com.br>
Date: Mon, 2 Jun 1997 22:22:07 -0300
To: <hfsig@tapr.org>
Message-ID: <199706030121.WAA12514@netra01.origin.com.br>
Rendered body type: text/plain

[Empty message body]

4. Re: unsubscribe till July....Please

From: Fred Percvial <save@io.com>
Date: Tue, 03 Jun 1997 10:56:07 +0000
To: hfsig@tapr.org
Message-ID: <3393F847.59F4@io.com>
Rendered body type: text/plain

I will be gone,
could you unsubscribe me.

5. Re: Unsubscribe

From: Fred Percvial <save@io.com>
Date: Tue, 03 Jun 1997 11:11:21 +0000
To: hfsig@tapr.org
Message-ID: <3393FBD8.6983@io.com>
Rendered body type: text/plain

[Empty message body]

6. New Subscriber

From: Antony Martin <anthony.martin@gecm.com>
Date: Wed, 11 Jun 1997 09:03:31 +0100
To: hfsig@tapr.org
Message-ID: <970611090331.27263@oas01.gmcl.co.uk>
Rendered body type: text/plain

Hi to all members of hfsig@tapr.org

I've just subscribed to this service.

If this post is successful I'll follow up with more info about
my identity & interests.

A.N. Martin

7. New member

From: Antony Martin <anthony.martin@gecm.com>
Date: Thu, 12 Jun 1997 21:59:28 +0100
To: hfsig@tapr.org
Message-ID: <970612215928.14171@oas01.gmcl.co.uk>
Rendered body type: text/plain

Hi! I'm Anthony. I just joined the hfsig@tapr.org .

I'm not a radio amateur, & I don't have a callsign.

I work for a commercial organisation in the field of
HF communications.

My expertise is in design of HF equipment, RF,
analogue & DSP hardware, and DSP software.

I have particular interests in HF modems as I've
designed them, programmed them & they're out there and
on the air at an HF station near you.

I have also done marine radiotelex (DSC distress
alerting) modems in use with HM coastguard around the UK.

I am currently working on a frequency hopping modem for
military radio and LPC-10 voice compression software.

As I'm not a ham, pardon me if I ask for clarification of
obtuse terminology or regulatory issues, although I think
I've got the gist of most of it.

Perhaps I'd cosider getting involved if it weren't for
a large obstruction called the morse test. I have no
interest in such low performance low data rate digital
modes. I think I have better uses for my brain than train
it to do the job of a DSP.

This means I'm not a user of HF which is a bit of a
disadvantage as I'm not as knowledgeable of the
conditions on HF as I'd like to be.

Personally I'm not interested in the current state of
amateur packet radio either; with the readily available
data bandwith of the internet, I personally wouldn't put
up an aerial for less than 64Kbit/s!

As the result of my work is the intellectual property of
the company, I have to set some limits to what I can
disseminate. I can't say exactly how our modems work,
or post you the sourcecode but what I can do is:
- - explain basic principles of operation. (Few modulation
    methods are new.)
- - answer questions like 'how could xyz be done'. (You can
    always ignore any advice.)

I also have access to Texas' DSP tools, emulators, simulators
and all that jazz, which is very useful (except the compiler).

As the hfsig looks like a place where wireless modems get
discussed, designed & a good going over, I thought I'd
drop in, learn something I don't know, offer some info &
generally be a helpful person.

- --
A.N. Martin
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

8. Re: [HFSIG:2128] New member

From: Marty Albert <marty@trucom.com>
Date: Thu, 12 Jun 1997 22:00:08 -0500
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970613030008.006769e4@mail.trucom.com>
Rendered body type: text/plain

At 16:08 06/12/97 -0500, Anthony wrote:
>
>Hi! I'm Anthony. I just joined the hfsig@tapr.org .

Hi, Anthony! Welcome!

>I'm not a radio amateur, & I don't have a callsign.

<snip>

Too bad.... The Ham ranks could use someone with your experience.

>Perhaps I'd cosider getting involved if it weren't for
>a large obstruction called the morse test. I have no
>interest in such low performance low data rate digital
>modes. I think I have better uses for my brain than train
>it to do the job of a DSP.

hehe! 'Nough said...

>Personally I'm not interested in the current state of
>amateur packet radio either; with the readily available
>data bandwith of the internet, I personally wouldn't put
>up an aerial for less than 64Kbit/s!

Stick around... With any luck, we'll be running faster than that on the
LANs, at least.

>As the hfsig looks like a place where wireless modems get
>discussed, designed & a good going over, I thought I'd
>drop in, learn something I don't know, offer some info &
>generally be a helpful person.

You might also want to look at the Network SIG and, perhaps, the BBS SIG...
Many other discussions go on there as well.

Take Care & 73

Marty Albert - marty@trucom.com
Amateur Radio: KC6UFM@KC6UFM.#SEMO.MO.USA.NOAM

****
Interested in Amateur Radio?
http://www.trucom.com/ppages/marty
****
HAmPS Coordinator
http://www.geocities.com/CapeCanaveral/Lab/7067
****

9. UUCP over HF

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 13 Jun 1997 10:02:26 +0100
To: hfsig@tapr.org
Message-ID: <33A10CA2.2E77@dircon.co.uk>
Rendered body type: text/plain

Hello,

Does anyone have any experience of running UUCP over an HF circuit.
I know that Nrad have immplemented an HF Email system based on JNOS
and a MIL-STD 188-110A serial tone modem.

I was wondering if UUCP would be more suitable than SMTP/TCP/IP ie
whether it has a lower overhead and less problems with turnaround
times as it was designed for slow dial-up connections. Also its dial
up nature seems suited to triggering an ALE controller to establish
the connection as UUCP can be configured to make outgoing connections
at scheduled times, hence giving more control over channel usage.

I notice however that UUCP seems to have quite a few security problems.

I don't particularly want to write my own MTA preferring to use a
Linux based one. Also it would nice to use a commercially available
UA.

At the moment this is just an investigation, however I do have
available the components to put together such as system. If this is
not considered a suitable subject for HFSIG please reply by private
Email.

Regards Charles G4GUO

10. Re: [HFSIG:2128] New member

From: RLANIER@mailb.harris.com (RLANIER)
Date: Fri, 13 Jun 1997 13:12:12 -0400
To: hfsig@tapr.org
Message-ID: <3a17f870@mailb.harris.com>
Rendered body type: text/plain

Hi Anthony,

Since you're doing a spread-spectrum design, you might want to subscribe to the SS list. We are currently discussing the new rule changes and submitting our opinions to the FCC.

There is also discussion about building a SS "device," whatever that may be - possibly an HF modem.

Since you have DSP experience, subscribing to the DSP list is another good idea. Sometimes really good discussions pop up.

I would encourage you to get your ticket (license). Don't let the code get in your way; learn it once and then forget it, if that's what you want to do.

I would definitely like to discuss SS modems with you someday.

Tony

_____ Reply Separator _____
Subject: [HFSIG:2128] New member
Author: hfsig@tapr.org at smtp
Date: 6/12/97 4:08 PM

Hi! I'm Anthony. I just joined the hfsig@tapr.org .

I'm not a radio amateur, & I don't have a callsign.

I work for a commercial organisation in the field of HF communications.

My expertise is in design of HF equipment, RF, analogue & DSP hardware, and DSP software.

I have particular interests in HF modems as I've designed them, programmed them & they're out there and on the air at an HF station near you.

I have also done marine radiotelex (DSC distress alerting) modems in use with HM coastguard around the UK.

I am currently working on a frequency hopping modem for military radio and LPC-10 voice compression software.

As I'm not a ham, pardon me if I ask for clarification of obtuse terminology or regulatory issues, although I think I've got the gist of most of it.

Perhaps I'd cosider getting involved if it weren't for a large obstruction called the morse test. I have no interest in such low performance low data rate digital modes. I think I have better uses for my brain than train it to do the job of a DSP.

This means I'm not a user of HF which is a bit of a disadvantage as I'm not as knowledgeable of the conditions on HF as I'd like to be.

Personally I'm not interested in the current state of amateur packet radio either; with the readily available data bandwith of the internet, I personally wouldn't put up an aerial for less than 64Kbit/s!

As the result of my work is the intellectual property of the company, I have to set some limits to what I can disseminate. I can't say exactly how our modems work,

or post you the sourcecode but what I can do is:
- - explain basic principles of operation. (Few modulation
    methods are new.)
- - answer questions like 'how could xyz be done'. (You can
    always ignore any advice.)

I also have access to Texas' DSP tools, emulators, simulators
and all that jazz, which is very useful (except the compiler).

As the hfsig looks like a place where wireless modems get
discussed, designed & a good going over, I thought I'd
drop in, learn something I don't know, offer some info &
generally be a helpful person.

- --
A.N. Martin
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

11. Re: [HFSIG:2131] Re: New member

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 13 Jun 1997 15:14:31 -0800
To: hfsig@tapr.org
Message-ID: <199706132201.PAA14991@peak.org>
Rendered body type: text/plain

Hello Anthony,

Welcome to HFSIG. You definately are at the right place.

>As the result of my work is the intellectual property of
>the company, I have to set some limits to what I can
>disseminate. I can't say exactly how our modems work,
>or post you the sourcecode but what I can do is:
>- - explain basic principles of operation. (Few modulation
>   methods are new.)
>- - answer questions like 'how could xyz be done'. (You can
>   always ignore any advice.)

That is OK with us.

How about sharing your views on contemporary modem design, i.e, massively
parallel vs. high speed single tone with equalization. Something like the
folks at DSTO have been working on.

Johan Forrer, KC7WW
HFSIG moderator (were still looking for a replacement!)

12. Re: [HFSIG:2128] New member

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Sat, 14 Jun 1997 11:26:06 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199706140826.LAA17244@boojum.hut.fi>
Rendered body type: text/plain

> Perhaps I'd cosider getting involved if it weren't for
> a large obstruction called the morse test. I have no
> interest in such low performance low data rate digital
> modes. I think I have better uses for my brain than train
> it to do the job of a DSP.

With your background you could do it. For a novice it is not that tough.
And it adds one more skill to your multi talent personality. I mean if
you take it as a challenge. And after you have the ablity to understand Morse
code you can just use it for getting a rough idea what is happening on the
band if nothing else. Some 15 years ago I was very keen on electronics.
At my home town I could not find a decent club for this hobby. Only the
radio amateurs offered well organized education and training. And I learned
also the Morse code even if it was not the original idea. I have not really
regretted that.

Jaakko/OH7BF

13. BlasterTeLetype V1.33 beta release

From: Rob Glassey <robglassey@geocities.com>
Date: Sat, 14 Jun 1997 16:53:19 -0700
To: hfsig@tapr.org
Message-ID: <33A32EEF.1678@geocities.com>
Rendered body type: text/plain

Hi Everyone!!

```
******************    A N N O U N C I N G    ******************

  ***    *****    *             *   *     *       ***    ***
  *  *     *      *             *   *     *          *      *
  ***      *      *              * *      *         **     **
  *  *     *      *              * *      *           *      *
  ***      *      ****             *      *   *   ***    ***

*************************  B E T A  *************************
```

Yes, BTL V1.33 Beta is now available via the WWW from my homepage:

http://www.geocities.com/SiliconValley/Heights/4477/

This is quite close to what was originally promised for V2.0,
and is well beyond what I in thought V1.5 would be.

I hope to finalise V1.5 in a week or two, so I'd very much appreciate
any feedback I can get from all you keen beta testers (and even a few
registered users out there too - Many thanks!! things are happening!).

If you do not have WWW access and would like to Beta test V 1.33
I can Email you a copy.

E N J O Y ! ! !

Cheers,

Rob, OH2/G0VTQ

robglassey@geocities.com
http://www.geocities.com/SiliconValley/Heights/4477/

P.S. Here's a brief summary of features, plus extracts from the help
that may give some insight.

```
               B T L   1.33  F E A T U R E S
              ~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~~
```

- High performance RTTY Reception and Transmision
- Function key TX buffers with simultaneous RX & editing.
- In-line text file transmission during TX mode.
- On screen tuning and audio level indicators
- Fully adjustable centre frequency and shift, with standard presets.
- User definable preset shift and center frequency
- Fully adjustable baud rate, 25 to 200 Baud
- USB or LSB default Mark/Space polarity for receive and transmit
- RXR Mark/Space tone reversal on receive
- Narrow/Wide tone filter option
- Unshift on Space (UNS)
- Extended Baudot, HAMCOMM compatible, plus US and ITU2 baudot standards
- Received text logging to a file
- 43/50 line mode
- Manually forced letters or figures shift
- Manually forced New Line and Clear Screen on demod screen
- Demodulation pause key (Hold)
- Input levels, tones, and many other parameters saved between sessions
- Support for Sound Blasters version 1.0 through AWE32
- Sound Blaster parameter validation
- On-Line HELP!!!
- Bright Colours!

Keys:
~~~~~

F1	HELP!!
ALT-X	Quit or Abort transmission
SPACE	Force letters mode
F	Force figures mode
Enter	New line
Home	Clear screen
I	Input level and selection
L	LOG open/close log file
H	Hold, Pause demodulation
S	USB/LSB mode
R	RXR Mark/Space reversal
U	UNS unshift on space toggle
N	NAR narrow filter mode
D	Display more, 43/50 lines
B	Select US/ITU2 Baudot standard
E	EXTended mode, auto detect
ALT-E	EXT mode (always on)
ALT-C	EXTF mode, fixed case
C	Change case (upper/lower)
Up/Down	Select preset centre freqs
PGUP/PGDN	Select preset shifts
ALT-P	Store Preset shift/centre freq
Arrows < >	Tune centre freq +/- 25 Hz
CTRL < >	Tune centre freq +/- 5 Hz
ALT < >	Tune centre freq +/- 1 Hz
CTRL-PG^/v	Variable shift +/- 10 Hz
ALT-PG^/v	Variable shift +/- 1 Hz
, or .	Reduce/increase baud rate by 5
< > or ; :	Reduce/increase baud rate by 50
ALT , or .	Reduce/increase baud rate by 1
ALT-Fn	Edit function key text
ALT-W	Word transmit mode toggle
T	TX mode
ESC	End TX mode
ALT-X	Abort transmission
ALT-B	Send Brag file in TX mode
F2-F12	Send Function key text in TX mode

C O N T E N T S
~~~~~~~~~~~~~~~~~

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14. Re: [HFSIG:2134] BlasterTeLetype V1.33 beta release

From: k5cnf1@juno.com (Richard B Morrow)
Date: Sat, 14 Jun 1997 22:00:14 EDT
To: hfsig@tapr.org
Message-ID: <19970614.205830.8023.14.k5cnf1@juno.com>
Rendered body type: text/plain

I am curious, will this work with other soundcards that emulate the
soundblaster family?

Richard, K5CNF

15. Re: [HFSIG:2134] BlasterTeLetype V1.33 beta release

From: Rob Glassey <robglassey@geocities.com>
Date: Sun, 15 Jun 1997 18:24:13 -0700
To: k5cnf1@juno.com, hfsig@tapr.org
Message-ID: <33A495BD.1C43@geocities.com>
Rendered body type: text/plain

Hi Richard,

> I am curious, will this work with other soundcards that emulate the
> soundblaster family?
>
> Richard, K5CNF

Yes, it should work with most card that emulate the Sound Blaster. I
have one such card on my PC at work and it seems to work fine.

BTL should work on virtually all Sound Blaster compatible cards,
although the TX side has been tested less than RX. I have a old cheap
and nasty 8 bit SB 2 compatible clone as well which it seems to work
fine on.

If anyone has problems with specific cards let me know, although in the
past trying to fix hardware problems without a sample of the hardware is
a difficult process, but it has been done.

To date I know of no SB compatible cards BTL does not work on.

The number one problem people have with the sound card is trying to run
BTL under Windows 3.X. It does work sometimes, but not always. If you
have problems QUIT WINDOWS.

Also some cards that emulate the SB need to be set up by loading some
driver or config program, these are often windows drivers, so windows
must be run first, then quit windows and run BTL.

Make sure the BLASTER enviroment variable is set correctly.

Cheers,

Rob

16. Re: [HFSIG:2132] Re: New member

From: Antony Martin <anthony.martin@gecm.com>
Date: Mon, 16 Jun 1997 22:02:05 +0100
To: hfsig@tapr.org
Message-ID: <970616220205.16061@oas01.gmcl.co.uk>
Rendered body type: text/plain

Hi to HFsig users - answering a few questions that were fired last week
- - This place suddenly woke up & deluged me with mail!

> Hello Anthony,

>
> How about sharing your views on contemporary modem design, i.e, massively
> parallel vs. high speed single tone with equalization. Something like the

> folks at DSTO have been working on.
>
>
> Johan Forrer, KC7WW
> HFSIG moderator (were still looking for a replacement!)
>

I have no personal views or opinions on the subject; which one's best
for you depends on your application. As ever, it comes down to
engineering choices. Of course, we all like to champion our own
products too.

I've personally done parallel (OFDM) modems, but the company sells
all sorts; the serial ones are done by a Marconi company in
austrailia. We also have mil-std ALE.

Often the choice may be dictated by other factors than simple
BER vs. SNR curves, or BER on a CCIR channel sim. Amateurs may be
using modems over the extreme ends of the SNR curves, because
they like low power work, or huge aerials or whatever, whereas
most commercial organisations may be more realistic.

Some points to consider:

 Receiver lock time:
Does your modem take milliseconds to sync up & produce good data?
Some take variable amounts of time to aquire. Could have a big
effect on half-duplex ARQ protocols.
One modem I know of potentially could take up to 4 seconds.
You can always throw error correction & large interleave at this
problem, but that will add even more delay...

 Frequency offset:
What offsets does it cope with?
Clover is reputed to tolerate only 10Hz. I think that's apalling. You
need at least 30Hz for practical applications (ie 1ppm) and probably
at least 100Hz for amateur use. Maybe your radios just aren't
accurate enough.

 Sync in mid-data:
If you're setting up a data broadcast facility, receivers have to be
able to aquire lock without seeing any special training sequence.
With some modulation methods, sync on random data is virtually
impossible or requires inordinate signal processing power.

 Traffic on-channel:
How do you tell the difference between a high-rate modem on-channel
and plain noise? AGC or Squelch is inadequate at low SNR. This is often
impossible or requires inordinate signal processing power. But very
important if you're setting up a radio data network, as you loose a
lot of traffic thru being stomped on by your own transmitters.

 Performance in phase noise:
See below.

 Performance thru radio IF filters:
Real radios have phase noise and chronic phase responses. Note that
channel simulators simulate the bit of the channel between the
transmit antenna & the receive antenna, but *not* the radios.
Some modulation methods in the research papers perform well on a
channel simulator, but badly over radios.

 Backwards compatability or conformance to a standard:
Often a good way to compromise performance but improves usability.
Usually the standard was written after a particular equipment which
was the first of it's type and the people who designed it were
total blockheads.

Don't simply judge modems by the underlying technology. Many of the
known problems of a given format can be 'got round' by simple
solutions. A different waveform for synchronisation, frequency
offset correction, differential encoding and things like this
can make a big difference.

Given that we look at a raw data rate like 2400bits/s (to avoid
getting dragged into the 2 bits/s morse debate) the parameters
of the parallel/serial modems are:

Currently, serial modems are available to 4800bits/sec and
parallel to 2400 bits/sec, although I wouldn't have much
problem doing a 4800bits/sec parallel modem. Of course,
no modem breaks Shannons' Law so you pay for your data
rate by requiring 3dB more SNR or take 3dB hit on your BER.

* Serial modems take a variable time to aquire equilisation.
 Parallel modems do not need to aquire equilisation.
 My parallel modem syncs in 200ms.

* Serial modems simply 'drop dead' at low SNR. Hence they
usually provide multitudes of lower rates. Parallel
modems work on down to 0dB SNR, often with a still-usable
error rate.

* Serial modems do not suffer degraded performance with
multipath.

* Parallel tone modems are unaffected by radio channel
filters but can be affected by poor close-in phase noise
of the radio.

* Parallel tone modems have higher peak-to-mean. Methods
can be used to reduce this to 4 - 5dB. As few transceivers
are continuously rated, 0dB peak-to-mean may not help.
Speech on a telephone is nominally 11dB peak-to-mean.
If compressed, on a 3kHz radio channel it can be 6dB
peak-to-mean, which is what most SSB transmitters are
designed for.

* serial modems require more processing power & more
memory to demodulate than parallel tone. My parallel
modem runs half-duplex in 1.2 Mips on a 20Mips TMS320C5x
DSP... embarassingly small.

* Someone appears to have now developed a frequency
- -hopping serial modem, although it is rumoured that the
limit for hopping serial modems is 5 hops/sec. We're doing
hopping at "greater than 20 hops sec" (for some stupid
reason no-one reveals actual hop rates) using parallel
modems at 2400 bits/sec and have been doing so since early
eighties... The plonkers in question are trying to make an
increadibly slow hopping modem into a NATO standard...
Is NATO being sold a load of old cobblers?

In my application, we put compressed encrypted speech over
modems so sync time & end-to-end delays are important.
As it's military, and you don't want the enemy DF'ing you
and shooting you up every time you press the PTT, so we
also frequency hop. The only choice here is parallel.

There is currently considerable interest in OFDM. OFDM is
specified for Digital Audio Broadcast & Digital TV Broadcast.
For DAB there's about a thousand carriers and the symbol rate
is about 1ms. That puts the data rate at 2 megabits. This
encodes about 4 stereo audio channels. It takes 4 Sharc DSPs
to do the trasmission encoding.

There is also a lot of interest in serial, with lots of the
major players now 'ignoring' parallel and going serial.
Its advantage of low degradation in the presence of
multipath appears to be the major criterion, although in
real applications like ARQ protocols other factors can
considerably reduce throughput of modems.

Must go, running out of time.
- ---
A.N. Martin
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- ---

17. unsubscribe

From: guillaume <guillaum@worldonline.nl>
Date: Mon, 16 Jun 1997 23:19:19 +0200
To: hfsig@tapr.org

Message-ID: <3.0.1.32.19970616231919.0068d248@pop3.worldonline.nl>
Rendered body type: text/plain

[Empty message body]

18. RE: [HFSIG:2138] unsubscribe

From: "Norman R. Johnson" <normj@interaccess.com>
Date: Mon, 16 Jun 1997 21:00:48 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BC7A98.5EA328A0@hp>
Rendered body type: text/plain

[Empty message body]

19. Re: [HFSIG:2137] Re: New member

From: dona@tridelta.com (Don Adams)
Date: Tue, 17 Jun 1997 08:41:51 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199706171241.IAA16734@sparcy.tridelta.com>
Rendered body type: text/plain

>
>I have no personal views or opinions on the subject; which one's best
>for you depends on your application.

>Some points to consider:
>
> Receiver lock time:

> Frequency offset:

> Sync in mid-data:

> Traffic on-channel:

> Performance in phase noise:

> Performance thru radio IF filters:

> Backwards compatability or conformance to a standard:

>* Serial modems take a variable time to aquire equilisation.

etc etc etc
>- ---
>A.N. Martin
>Development Engineer
>GEC-Marconi Communications Ltd.
>These may or may not be the opinions of my employer.
>- ---

This overview was *great*. It's really good to talk to someone
with such practical experience.....

over....Don wa8qzz

20. RE: [HFSIG:2137] Re: New member

From: jalocha@axdele.cern.ch
Date: Tue, 17 Jun 1997 16:37:47 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199706171437.QAA30771@dxmint.cern.ch>
Rendered body type: text/plain

> My parallel modem syncs in 200ms.

This is at some pre-determined pattern or anywhere in the middle
of a transmition ?

I ask, because for my parallel-tone modem getting the synchronization
and tuning in the middle of a transmition looks like an impossible
task. I didn't even work out how to keep up the symbol-sync.
(but keeping th etuning is easy) during transmition
so I only acquire it at the preamble and hope
the transmition is short enough (these are AX.25 packets)
so the Tx and Rx clocks are accurate enough.

Pawel

21. Re: [HFSIG:2141] Re: New member

From: Antony Martin <anthony.martin@gecm.com>
Date: Tue, 17 Jun 1997 18:54:22 +0100
To: hfsig@tapr.org
Message-ID: <970617185422.5420@oas01.gmcl.co.uk>
Rendered body type: text/plain

> > My parallel modem syncs in 200ms.
>
> This is at some pre-determined pattern or anywhere in the middle
> of a transmition ?
>
> I ask, because for my parallel-tone modem getting the synchronization
> and tuning in the middle of a transmition looks like an impossible
> task. I didn't even work out how to keep up the symbol-sync.
> (but keeping th etuning is easy) during transmition
> so I only acquire it at the preamble and hope
> the transmition is short enough (these are AX.25 packets)
> so the Tx and Rx clocks are accurate enough.
>
> Pawel
>

We have a predefined sync sequence at the start of messages. Reduce the number of tones so frequency offsets don't cause a problem. You also need to get your frequency offset close very quick too, so try and get frequency info from the sync sequence.

Symbol sync is easy: Use the standard method used in most modems. It's variously called the sliding window or the early-gate late-gate method.

You do 3 demodulations of the data at slightly different times. An early one, a centre one, and a late one. You must then derive a measure of demodulation performance or quality value for each of the demodulations. If the early one demodulates better than the late one, advance the timing a bit. If the late one is best, retard it.

No prizes for guessing that the middle demodulation supplies the data. You could choose the one with best demod to supply output data, it depends what you think. I found that occasionally my quality value got 'fooled' and output duff data when I did that.

Getting sync on the data is indeed very difficult with almost all high-rate modem waveforms. I've seen papers on how to do this with equilised trellis coded modulation; and, yes, the conclusion was that its very hard and impossible to guarantee it'll work in any given situation.

You would think that by doing enough demodulations at different timings you would be able to pick the one that demodulated best and eventually decide if you had a real incoming modem signal. Yes it takes a lot of processing, but if a modem takes 1Mips, not much problem.

In practice the frequency offset virtually scrambles things and without any idea of frequency offset or symbol timing makes the task extremely difficult and very computation and memory intensive.

But it has almost been done - we can recognise one of our own modems on the channel and give a positive indication (% confidence) within about 1.5 seconds. In doing this it did obtain symbol timing and did determine the incoming frequency error. It is only a small step from there to data demodulation but we were not really interested at the time. In this application the station couldn't make use of data from half-a-transmission, but it could move on & look for a free channel to transmit on, which increases the throughput of the system.

Simultaneously with the above you have to keep looking for normal start-of-message which adds to the processing.

We threw almost all of 20 Mips in carefully optimised 'C50 code at the problem to do this. It can be done.

- --
A.N. Martin
Development Engineer
GEC-Marconi Communications Ltd.

These may or may not be the opinions of my employer.
- --

22. Serial versus parallel, another advantage for serial

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Tue, 17 Jun 1997 23:18:38 -0400
To: hfsig@tapr.org
Message-ID: <33A7538E.2D54@worldnet.att.net>
Rendered body type: text/plain

Antony Martin wrote:
>
> > > My parallel modem syncs in 200ms.
> >
> > This is at some pre-determined pattern or anywhere in the middle
> > of a transmition ?
> >
> > I ask, because for my parallel-tone modem getting the synchronization
> > and tuning in the middle of a transmition looks like an impossible
> > task. I didn't even work out how to keep up the symbol-sync.
> > (but keeping th etuning is easy) during transmition
> > so I only acquire it at the preamble and hope
> > the transmition is short enough (these are AX.25 packets)
> > so the Tx and Rx clocks are accurate enough.
> >
> > Pawel
> >
>
> We threw almost all of 20 Mips in carefully optimised 'C50 code at
> the problem to do this. It can be done.
>
> - --
> A.N. Martin
> Development Engineer
> GEC-Marconi Communications Ltd.
> These may or may not be the opinions of my employer.
> - --

This is one great advantage for serial modems with their long preamble
sequences providing for cross correlation with known waveform which
allows for timing acquisition in all but the worst multipath conditions.
Even in the worst multipath conditions, the Rohde & Schwarz, Mil Std 188
and Stanag 4285 modems can have preamble acquisition by a more
computationally intensive algorithm involving a relationship between
symbols that are on a stride of 32 symbols apart. This is because all
of
them use a 32 symbol "preamble randomizer" which is repeated and the
underlying data is "bpsk" (symbol 0 and 4) sending Walsh functions. The
relationship between the complex signal samples 32 baud times apart is
easily derived but computationally expensive to compute on the fly.
I can mark preambles for all of the above modems (most especially the
Mil Std variants in the either the long or short inteleaver mode) when
the signal is barely audible because of the strong relationship. This
detection method is nearly insensitive to frequency offset and works
in just huge multipath primarily because even when the signal looks not
much like the original square root raised cosine filtered complex
symbols
with roll off factor 0.2 it still maintains its AUTOcorrelation
properties.
In my mudder demods, I use this to mark the place, find the frequency
that
does the best cross correlation job, find the FSE plus decision feedback
EQ that works on the known symbols and then all seems right with the
world.
You then begin spinning the known symbols into the right places (and the
feedback part of the equalizer does most of this) and then start
computing
factor averaged bit values from the observed symbols and put them into
the
right places in the deinterleaver matrices and then feed them to your
FEC decoder subroutine and out pops data. Seems easy enough eh? ;-).

When Harris, err uhh, I mean the government wrote Mil Std 188, Harris
patented "data directed demodulation". I haven't bothered to even
justify
this patent with a look but I bet it contains most of what I have said
since it is all so obvious.

Bob
--
****
* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
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* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

23. Re: [HFSIG:2134] BlasterTeLetype V1.33 beta release

From: Claude Frantz <claude@bauv106.bauv.unibw-muenchen.de>
Date: Wed, 18 Jun 1997 06:58:00 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199706180458.GAA16028@bauv106.bauv.unibw-muenchen.de>
Rendered body type: text/plain

According to Rob Glassey:
>
> Yes, BTL V1.33 Beta is now available via the WWW from my homepage:
>
> http://www.geocities.com/SiliconValley/Heights/4477/
>
> o High performance RTTY Reception and Transmision
> o Function key TX buffers with simultaneous RX & editing.

[.....]

Can you give us more information about the internals ?
Which filter functions are used ? How are they switched, etc...

Thanks.

73,
Claude

24. Re: [HFSIG:2143] Serial versus parallel

From: Antony Martin <anthony.martin@gecm.com>
Date: Wed, 18 Jun 1997 16:12:01 +0100
To: hfsig@tapr.org
Message-ID: <970618161201.15354@oas01.gmcl.co.uk>
Rendered body type: text/plain

> Antony Martin wrote:
> >
> > > > My parallel modem syncs in 200ms.
> > >
> > > This is at some pre-determined pattern or anywhere in the middle
> > > of a transmition ?
> > >
> > > I ask, because for my parallel-tone modem getting the synchronization
> > > and tuning in the middle of a transmition looks like an impossible
> > > task. I didn't even work out how to keep up the symbol-sync.
> > > (but keeping th etuning is easy) during transmition
> > > so I only acquire it at the preamble and hope
> > > the transmition is short enough (these are AX.25 packets)
> > > so the Tx and Rx clocks are accurate enough.
> > >
> > > Pawel
> > >
> >
> > We threw almost all of 20 Mips in carefully optimised 'C50 code at
> > the problem to do this. It can be done.
> >
> > - --
> > A.N. Martin
> > Development Engineer
> > GEC-Marconi Communications Ltd.
> > These may or may not be the opinions of my employer.
> > - --
>

Bob wrote:
> This is one great advantage for serial modems with their long preamble
> sequences providing for cross correlation with known waveform which
> allows for timing acquisition in all but the worst multipath conditions.
> Even in the worst multipath conditions, the Rohde & Schwarz, Mil Std 188
> and Stanag 4285 modems can have preamble acquisition by a more
> computationally intensive algorithm involving a relationship between
> symbols that are on a stride of 32 symbols apart.

<details snipped>

Yes I use substantial preamble sequence too. To a serial modem it's
a lot of symbols; but for a parallel modem it's a few. Serial
modems must average a lot of symbols to reduce the detection BW;
parallel modems need only average a few. The correlation of the
preamble still achieves the same result in both cases.

We can get sync on SNRs which are useless for traffic, although we try
to keep it sensible as this isn't a useful facility. The basic
preamble search on a parrallel modem takes very little processing.

But the whole problem in question is how do you get sync in mid-traffic
say in a broadcast or streaming application or when scanning channels
to identify if it is in use?

If you keep putting massive preambles in traffic this will reduce
throughput.

If you can't train an equaliser without "data direction" and without
equalisation _and_ symbol timing _and_ frequency offset you don't have
valid data. You don't stand a cat in hells chance of getting sync
on a serial modem without seeing a training sequence. Plus if
you propose modems that require numerous rates to work a decent
range of SNR you must recognise all of them.

Long interleave delays reduce troughput in practical applications.

> When Harris, err uhh, I mean the government wrote Mil Std 188, Harris
> patented "data directed demodulation". I haven't bothered to even
> justify
> this patent with a look but I bet it contains most of what I have said
> since it is all so obvious.
>
>

I'm very suprised that data directed equiliser updating is patented;
when was this patented? People use it all over the place; it's
textbook stuff. This is how (almost) all equilisers are adapted...
In everything from telecoms, satellite, troposcatter the whole
gamut. Can this be confirmed? Was its use patented specifically
in relation to HF modems of a certain kind?

The handbook for a certain Harris HF modem says in the specifications
"Equilisation adaption: data directed (patented)"

This is as clear as mud; it doesn't necessarily mean that all data
directed adaption of equalisers is patented; it does indicate that
some part of the method they use in this application _may_ be
patented in one or more countries.

Why are they afraid to indicate the patent numbers, dates & countries
so that people can get a copy and check? If someone can give me a
reference, I'll obtain a copy. As patents are public knowledge, it
should be legal for me to turn it into computer document & post
it here.

There are other kinds of equiliser adaption that use other
properties of the waveform to derive the error signal. In some cases
these perform better at low SNR:
Eg. See "A New Aproach to Multipath Correction of Constant Modulus
Signals", John R. Treichler & Brian G. Agee, ASSP-31,No.2, April 1983.

pps. I forgot to mention in previous postings that parallel modems
can operate with no degradation in the presence of significant
multipath. They don't necessarily adopt strict Orthogonal FDM, in
which any path delay degrades the BER vs. SNR curve.

I'm not anti-serial or anything; it doesn't offer what's needed in
the applications Ive worked on.

My assertion is that serial modems are unsuitable for military
HF tactical comms as they cannot operate with frequency-hopping,
whereas parallel modems can.

I also assert that parallel modems can offer features which are
beneficial for automatic HF managed networks. By preventing lost
traffic from collisions it prevents the early collapse of the
network as data throughput increases.

I also assert that serial modems are ideal for point-to-point
file transfer and point-to-point permanent duplex links.

If anyone knows of any papers on the comparative performance of
serial modems in jamming I'd like to know of them, as this appears
to be an unknown area.

I'd like to do a serial modem one day though. The 8 Megabits
troposcatter ones look quite interesting. (We do those here too.)
Is there such thing as amateur tropo?

- --
A.N. Martin
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

25. NEWQPSK news from Finland

From: jalocha@axdele.cern.ch
Date: Wed, 18 Jun 1997 17:22:55 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199706181522.RAA29527@dxmint.cern.ch>
Rendered body type: text/plain

With Timo's permission I pass to the HFSIG the (not that bad) news
Finland on the NEWQPSK tests.

Pawel

==========

Timo writes:

latest news regarding the Finnish NEWQPSK tests:

The test forward between Espoo and Kouvola gave some very concrete results:

1. It is not very good to have the HF rig of a club station connected to the
 local packet BBS. People tend not to understand why the HF rig should be
 left on. As a result the Kouvola end of the link was moved to the contest
 QTH of the club, in Kuusankoski. Unfortunately the QSO did not work very
 well from there. Reasons are not yet discovered. As long as the link was
 between the normal club qth the forward worked well. At the moment the link
 is again more or less active daily on 3591.4 kHz.

2. Ville and the guys from his club were on a DXpedition to Kökar, an island
 that belongs to the Åland islands. They brought the DSP with them and were
 accessing the packet network via Joni's home station QPSK gateway. I worked
 them almost every evening. They were using the call OH0LIZ on 3591.4 LSB
 FEC3. QSO was very good early in the evenings but during nights the QRM
 became so bad that nothing got trough. Joni might have some more info
 regarding the reliability during daytime. Anyway, OH0LIZ was all the time
 logged on to the Finnish DXcluster network over a 200-300 km NEWQPSK link.

3. There is something wrong with the carrier detection of the modem. If
 there are more than one station on the same frequency, there will be a lot
of collisions, a lot more than in normal packet qsos. The modem transmits on
 top of another transmission very often. Is there a possibility that the
 modem does not understand the preamble as a proper signal and simply
 transmits on top of it? Or is it so, that the modem believes that the
 transmission ends after the preamble, and decides to transmit? At the moment
 this is the worst problem.

4. TCP/IP. I have now switched over to TCP/IP over QPSK. I have an old

```
 486/33 PC at home which is not in any sensible use, so I downloaded the
 latest JNOS version 111x7 and after some testing finally got on air. It took
 me some time to learn how to set the KISS parameters, especially as JNOS of
 some reason seems not to support all possible parameters. I had to use the
KISS parameter that NOS labels Hardware to change FEC. Now changing FEC
 coding works very well, but we did not manage to get the interleave factor
to change by using a KISS parameter. We tried it so, that we created a
 connection, and then the other tried to change his interleave. Nothing
 changed, the QSO still worked. We concluded that the KISS parameter did not
 affect the interleave factor.

5. Modem speed. TXtuneLen and TXSyncLen are working very interestingly. We
tried to reduce the transmission overhead, once again, by shortening the  TX
 and RX parameters related to the preamble. The interesting finding was, that
 with SampleFreq 9600 and FEC 3 we can  use TXtune and Sync len 22 without
 any problems. With SampleFreq 8000 it will not work anymore with those
 settings. So in practice this means that we have defaulted to samplefreq
 9600 and FEC3.

6. Future plans: I think the next step will be to see why the modem does not
 work OK at the contest QTH when the boys come back from OH0 land. We have
 also been asked to write an article about our experiments to Radioamatööri,
 the magazine of SRAL, the Finnish amateur radio league. I think that might
be something we will also concentrate on in the future. There have been some
 preliminary discussions with SRAL of SRAL subsidizing a few HF NEWQPSK links
 to support the finnish packet backbone network. This article would be the
 introduction to this project, a sort of a test to see if there would be
 interested amateurs and clubs to participate in the link building. You don't
 happen to have a picture of yourself in JPEG format that we could use in the
 article?


Best regards, CUQ (CU on QPSK)

P.S Be sure to listen to Radio Hami on 1584 kHz during the summercamp of
 SRAL.

Timo

==========================================================================
```

26. Modern Modems

From: Bill Farmer <fxfarmer@erols.com>
Date: Wed, 18 Jun 1997 11:35:11 -0400
To: hfsig@tapr.org
Message-ID: <33A8002F.337@erols.com>
Rendered body type: text/plain

```
Can anyone put a name on the type modem that is producing the signal I
hear on 10,760 kHz USB ?

Have also heard this on 9225, 11425, and 14772 kHz.  My curiosity is
killing me.

Thanks, Bill    W3CSW
        Rockville, MD
```

27. Re: [HFSIG:2147] Modern Modems

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 18 Jun 1997 16:26:48 -0800
To: hfsig@tapr.org
Message-ID: <199706182326.QAA12155@peak.org>
Rendered body type: text/plain

```
Bill,

>Can anyone put a name on the type modem that is producing the signal I
>hear on 10,760 kHz USB ?
>
>Have also heard this on 9225, 11425, and 14772 kHz.  My curiosity is
>killing me.
>
>Thanks,        Bill    W3CSW
>       Rockville, MD
>
>

I picked up a signal on 10,760 USB, but not sure that it was what you were
referring to. It is a simplex signal, comes on and stays on for ever. There
```

may be a reverse channel somewhere else. Sounds like a single tone m-PSK,
symbol rate > 100 baud. If you listen carefully, there is a constant pilot
tone at some reduced level that I presume are used for doppler.

Is that it?

--Johan

28. Re: [HFSIG:2145] Re: Serial versus parallel

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 18 Jun 1997 16:26:49 -0800
To: hfsig@tapr.org
Message-ID: <199706182326.QAA12160@peak.org>
Rendered body type: text/plain

Anthony,

Interesting thoughts...

>pps. I forgot to mention in previous postings that parallel modems
>can operate with no degradation in the presence of significant
>multipath. They don't necessarily adopt strict Orthogonal FDM, in
>which any path delay degrades the BER vs. SNR curve.

I assume that you mean relative low symbol rates, i.e., below 100 baud.

Do you also advocate the use of guard bands in the OFDM demodulation
process, i.e.,
instead of using all the data from the entire symbol time, the FFT process only
uses a slice out of the middle of the baud, thus leaving a guard band each
side to
absorb multipath jitter?

Continuing on the issue of OFDM; Some of us have been playing around with
RC pulse shaping applied to individual carriers within the OFDM ensamble,
only to find that one then needs wider spacing the 1/T orthogonal spacing. The
explanation for this, I only found recently in Mischa Schwartz's book
(Information Transmission, Modulation, and Noise ISBN 0-07-055782-9, Section
3-9).
This clearly explains that true RC filtering doubles the bandwidth required.
Does this mean that most OFDM implementations do guard filtering of the
entire OFDM ensamble rather than filtering each individual tone component?
Probably makes sense.

--Johan

29. RE: [HFSIG:2149] Re: Serial versus parallel

From: jalocha@axdele.cern.ch
Date: Thu, 19 Jun 1997 15:20:49 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199706191320.PAA27370@dxmint.cern.ch>
Rendered body type: text/plain

Hello,

>Continuing on the issue of OFDM; Some of us have been playing around with
>RC pulse shaping applied to individual carriers within the OFDM ensamble,
>only to find that one then needs wider spacing the 1/T orthogonal spacing.

Yes, in my modem I shape the symbols as nice as I can and I have
to space the carriers at 1.5 of what the 1/T limit is, so I use 1.5 more
bandwidth (but not twice as much as Johan suggests). It is not trivial
to find the proper shape so the inter-symbol interferience (almost)
cancels at selected points. I think I figured out today how to squeeze
the symbols more so my "extra bandwidth" factor would go down to 1.3
or possibly 1.2.

I think that nicely shaping the symbols is a proper way - for example
you gain more resistance against CW carriers. If symbols are just squares
then a carrier in the passband is affecting all of them. When you make
the symbols smooth (=> more narrow-banded) a single carrier affects only the
ones which are close in frequency.

Pawel

30. unsubscribe

From: guillaume <guillaum@worldonline.nl>
Date: Wed, 18 Jun 1997 21:31:18 +0200

To: hfsig@tapr.org
Message-ID: <3.0.1.32.19970618213118.006919b0@pop3.worldonline.nl>
Rendered body type: text/plain

At 22:22 17-06-97 -0500, you wrote:
>Antony Martin wrote:
>>
>> > > My parallel modem syncs in 200ms.
>> >
>> > This is at some pre-determined pattern or anywhere in the middle
>> > of a transmition ?
>> >
>> > I ask, because for my parallel-tone modem getting the synchronization
>> > and tuning in the middle of a transmition looks like an impossible
>> > task. I didn't even work out how to keep up the symbol-sync.
>> > (but keeping th etuning is easy) during transmition
>> > so I only acquire it at the preamble and hope
>> > the transmition is short enough (these are AX.25 packets)
>> > so the Tx and Rx clocks are accurate enough.
>> >
>> > Pawel
>> >
>>
>> We threw almost all of 20 Mips in carefully optimised 'C50 code at
>> the problem to do this. It can be done.
>>
>> - --
>> A.N. Martin
>> Development Engineer
>> GEC-Marconi Communications Ltd.
>> These may or may not be the opinions of my employer.
>> - --
>
>This is one great advantage for serial modems with their long preamble
>sequences providing for cross correlation with known waveform which
>allows for timing acquisition in all but the worst multipath conditions.
>Even in the worst multipath conditions, the Rohde & Schwarz, Mil Std 188
>and Stanag 4285 modems can have preamble acquisition by a more
>computationally intensive algorithm involving a relationship between
>symbols that are on a stride of 32 symbols apart. This is because all
>of
>them use a 32 symbol "preamble randomizer" which is repeated and the
>underlying data is "bpsk" (symbol 0 and 4) sending Walsh functions. The
>relationship between the complex signal samples 32 baud times apart is
>easily derived but computationally expensive to compute on the fly.
>I can mark preambles for all of the above modems (most especially the
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>the signal is barely audible because of the strong relationship. This
>detection method is nearly insensitive to frequency offset and works
>in just huge multipath primarily because even when the signal looks not
>much like the original square root raised cosine filtered complex
>symbols
>with roll off factor 0.2 it still maintains its AUTOcorrelation
>properties.
>In my mudder demods, I use this to mark the place, find the frequency
>that
>does the best cross correlation job, find the FSE plus decision feedback
>EQ that works on the known symbols and then all seems right with the
>world.
>You then begin spinning the known symbols into the right places (and the
>feedback part of the equalizer does most of this) and then start
>computing
>factor averaged bit values from the observed symbols and put them into
>the
>right places in the deinterleaver matrices and then feed them to your
>FEC decoder subroutine and out pops data. Seems easy enough eh? ;-).
>
>When Harris, err uhh, I mean the government wrote Mil Std 188, Harris
>patented "data directed demodulation". I haven't bothered to even
>justify
>this patent with a look but I bet it contains most of what I have said
>since it is all so obvious.
>
>
>
>Bob
>--
>***
>* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
>Astronomy*

>* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
>*
>* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
>*
>* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
>*
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>*
>* (W) 609-279-6240   * Council. Brownie Troop 193
>*
>**
>
>
>

31. Re: [HFSIG:2149] Re: Serial versus parallel, mainly parallel

From: Antony Martin <anthony.martin@gecm.com>
Date: Fri, 20 Jun 1997 13:06:59 +0100
To: hfsig@tapr.org
Message-ID: <970620130659.11220@oas01.gmcl.co.uk>
Rendered body type: text/plain

> Anthony,
>
> Interesting thoughts...
>
> >pps. I forgot to mention in previous postings that parallel modems
> >can operate with no degradation in the presence of significant
> >multipath. They don't necessarily adopt strict Orthogonal FDM, in
> >which any path delay degrades the BER vs. SNR curve.
>
> I assume that you mean relative low symbol rates, i.e., below 100 baud.
>
I thought all parallel modems (for HF) operated with low symbol rates;
that's the whole idea. Our data rate 2400 bits/sec; the symbol rate
is less than 50 symbols/sec (frames/sec).

What rates are other modems using?

(I try not to use 'baud' as not everybody properly interprets it as
symbols/sec. If I said I had a 50 baud modem most people get the wrong
end of the stick, especially digital engineers...You get this incredible
look of horror & dismay... They say something like "can't you do better
than that on HF' Then you tell them it's 2400 bits/sec. The expression
changes to a blank look or one that says 'you're a nutter'. *smile*)

> Do you also advocate the use of guard bands in the OFDM demodulation
> process, i.e.,
> instead of using all the data from the entire symbol time, the FFT process only
> uses a slice out of the middle of the baud, thus leaving a guard band each
> side to
> absorb multipath jitter?

Yes, I advocate use of guard bands as you describe them. The guard
bands must be there when you transmit, you can't add them at the
receiver; of course the FFT won't work on less than a whole symbol.

>
> Continuing on the issue of OFDM; Some of us have been playing around with
> RC pulse shaping applied to individual carriers within the OFDM ensamble,
> only to find that one then needs wider spacing the 1/T orthogonal spacing. The
> explanation for this, I only found recently in Mischa Schwartz's book
> (Information Transmission, Modulation, and Noise ISBN 0-07-055782-9, Section
> 3-9).
> This clearly explains that true RC filtering doubles the bandwidth required.

Are you not effectively summing the signal in the symbol period at the
receive end? As integrate-and-dump is a classical way to apply the
raised-cosine response, does each tone not already have the optimal
raised cosine response? If you look at the frequency response of one
bin of an FFT, what does it look like? A bandpass raised-cosine.

So there's no point adding RC filtering individual tones?

And you're right about the BW - the modulation spectra of each
tone in an OFDM signal _DOES_ overlap the adjacent one, and hence in
theory the tones should be twice as far apart. Fortunately the
'Orthogonal' bit allows it to be all sorted out again.

Stop thinking about it too hard; It all works fine the way it is!
If it ain't broke, don't fix it.

Unless your BER is worse than predicted - you probably have some other
problem. What's the phase stability like over a symbol period on Ham
radio rigs? Our radios here are the dog's bollocks. I know, I designed
them. (Slight exaggeration; I did the RF of the recent ones.)

> Does this mean that most OFDM implementations do guard filtering of the
> entire OFDM ensamble rather than filtering each individual tone component?
> Probably makes sense.
>
Well, you need enough filtering to stop frequencies outside the tone set
ending up in one of the wanted FFT bins. If you don't it degrades SNR.

> --Johan
>

It may help to shape the start and end of symbols in the time domain in
the guard band; it may reduce the effect of multipath delay;
I'll think about this idea...

You haven't said what modulation you used on the tones; I take it it's
PSK or QPSK... I've not hard of anybody doing anything else.

(Unless you consider 8-ary to be 8-tone FDM with limited codeset
On-Off Keying?!!! *he he*)

Bi Bi
- --
A.N. Martin
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

32. Re: [HFSIG:2152] Re: Serial versus parallel, mainly parallel

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 20 Jun 1997 07:39:24 -0800
To: hfsig@tapr.org
Message-ID: <199706201439.HAA25924@peak.org>
Rendered body type: text/plain

Anthony,

>(I try not to use 'baud' as not everybody properly interprets it as
>symbols/sec. If I said I had a 50 baud modem most people get the wrong
>end of the stick, especially digital engineers...You get this incredible
>look of horror & dismay... They say something like "can't you do better
>than that on HF' Then you tell them it's 2400 bits/sec. The expression
>changes to a blank look or one that says 'you're a nutter'. *smile*)

You are amongst friends here - we understand that bauds and BPS may or may
not mean the same :-).

>Yes, I advocate use of guard bands as you describe them. The guard
>bands must be there when you transmit, you can't add them at the
>receiver; of course the FFT won't work on less than a whole symbol.

Interesting - I have seen it implemented with gaurd bands in the receiver
side only. It does, however, become is tricky demodulation issue when one
use differential phase encoding as then one have to apply a "correction"
factor to your phases as you move along.

>Are you not effectively summing the signal in the symbol period at the
>receive end? As integrate-and-dump is a classical way to apply the
>raised-cosine response, does each tone not already have the optimal
>raised cosine response? If you look at the frequency response of one
>bin of an FFT, what does it look like? A bandpass raised-cosine.
>
>So there's no point adding RC filtering individual tones?
>

I'm not sure I follow the gist. Doesn't RC filtering works with 1/2 part
done at transmit time and then the other 1/2 at receive time, i.e., we
filter the transmit side with root raised cosine (RRC) response, then filter
with a RRC response at the receive end too - that effectively produces the
needed RC response?

>And you're right about the BW - the modulation spectra of each
>tone in an OFDM signal _DOES_ overlap the adjacent one, and hence in
>theory the tones should be twice as far apart. Fortunately the
>'Orthogonal' bit allows it to be all sorted out again.
>
>Stop thinking about it too hard; It all works fine the way it is!
>If it ain't broke, don't fix it.

Yes, you are right - one should not worry about it, because it works that
way. However, this is not quite the issue. Have you looked at the Pactor II
waveform as an example? (see: http://www.SCS.com). The point that I am
trying to make is that; sometimes, for some odd reason, we need parallel
tones, not strictly orthogonal tones - those may be two different things.

>
>Unless your BER is worse than predicted - you probably have some other
>problem. What's the phase stability like over a symbol period on Ham
>radio rigs? Our radios here are the dog's bollocks. I know, I designed
>them. (Slight exaggeration; I did the RF of the recent ones.)

Ham radio equipment probably is not perfect either, but if we can handle the
challenge of the ionosphere successfully, I have hopes that we can tackle
the radios next :-).

>It may help to shape the start and end of symbols in the time domain in
>the guard band; it may reduce the effect of multipath delay;
>I'll think about this idea...

This is what I have seen implemented on some MIL-STD-188 parallel tone modems.

>
>You haven't said what modulation you used on the tones; I take it it's
>PSK or QPSK... I've not hard of anybody doing anything else.

Our experimental efforts are using m-DQPSK. Phil Karn (the famous KA9Q - Mr.
TCP/IP) is suggesting the offset form (SQPSK). There also are some efforts
using m-FSK, but PSK and PAM is also used for some VHF/UHF terrestrial and
satellite-based amateur radio work. Our biggest need for improvent on HF is
to improve our poorly performing 300 baud(300 BPS, binary FSK), AX25 CSMA
capabilities.

Till later,

--Johan

33. Re: [HFSIG:2153] Re: Serial versus parallel, mainly parallel

From: Antony Martin <anthony.martin@gecm.com>
Date: Fri, 20 Jun 1997 18:40:08 +0100
To: hfsig@tapr.org
Message-ID: <970620184008.15716@oas01.gmcl.co.uk>
Rendered body type: text/plain

> >
> >Unless your BER is worse than predicted - you probably have some other
> >problem. What's the phase stability like over a symbol period on Ham
> >radio rigs? Our radios here are the dog's bollocks. I know, I designed
> >them. (Slight exaggeration; I did the RF of the recent ones.)
>
>
> Ham radio equipment probably is not perfect either, but if we can handle the
> challenge of the ionosphere successfully, I have hopes that we can tackle
> the radios next :-).
>

My Apologies: The phrase 'dogs bo**ocks' where I work implies very good,
excellent, A1, the bees knees, the last word ...

Apparently it didn't translate very well!!! I shall avoid it & other
colloquisms in email in future. Knuckles proverbially rapped.

To reiterate: the HF equipment we sell is very high performance.

- --
A.N. Martin
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

34. Re: [HFSIG:2153] Re: Serial versus parallel, mainly parallel

From: Antony Martin <anthony.martin@gecm.com>
Date: Fri, 20 Jun 1997 21:28:24 +0100
To: hfsig@tapr.org
Message-ID: <970620212824.15716@oas01.gmcl.co.uk>
Rendered body type: text/plain

Hello all!
>
> I'm not sure I follow the gist. Doesn't RC filtering works with 1/2 part
> done at transmit time and then the other 1/2 at receive time, i.e., we
> filter the transmit side with root raised cosine (RRC) response, then filter
> with a RRC response at the receive end too - that effectively produces the
> needed RC response?
>
I've been trying to remember my communications lectures on raised
cosine and I've looked it up in my textbooks. The comments which follow
apply to comms systems in general, I'm not suggesting anything about
parallel OFDM modems.

There appear to be some misconceptions about it. My book doesn't
devote much space to it. There are a family of raised-cosine
characteristics; not one. One member of the family is the
well-known 'full-cosine' one. The brick-wall low-pass filter is
another. I didn't know that. Between these extremes there is a
range of responses, determined by the 'excess bandwith' factor,
which ranges from 0 to 1.

These raised-cosine charateristics are in turn part of another
family of responses called the nyquist waveforms which share
the charateristics of zero ISI.

The main advantages of the full-cosine is that it's slopes
are most gentle making it easily realizable. Some kind of
filtering is obviously necessary so why not choose the
easiest? But is it the best?

The full-cosine has the disadvantage that it increases
the spectrum used by a factor of 2 over that actually necessary.
Excess bandwith is a bad thing on HF as you have close interferers
and there's a bandwidth shortage.

As you've got a DSP, the complexity associated with high-order
filters is not a problem. This suggests that the steepest raised-
cosine should be chosen that still behaves itself with the
number of coefficients you're prepared to use. This is what you
get anyway if you design a FIR filter of a given order which
approximates brick-wall, with the 6dB BW equal to 1/(symbol rate).
Use 3dB BW as you want one at each end & the raised-cosine is the
overall channel response. (Suprise-Suprise!)

A quote from "Intro to Comms Systems", F. G. Stremler:
"If the effects of both ISI & additive noise must be considered,
then the transmitter and receiver filters each should have
magnitude frequency transfer functions proportional to the
square root of the desired spectral density at the output of
the receiver filter and inversely proportional to the square
root of frequency transfer function of the channel."

The Tx & Rx filters should be the same. (Matched filters.)

Note also that it applies only for gaussian (flat) noise.
This assumption may not be valid on HF. It's as good an
assumption as any.

Note that if you want to equalise the channel, half the

equalisation must be placed at each end. In serial modems,
the adaptive equalisation is at one end. This will result in
an SNR penalty.

- -----

? I couldn't find anything remotely HF on www.SCS.com

- -----

I do see your point re. large-signal CW interferers.
If the CW is similar to your own tones, it should only blat
one of the parallel tones. If it's much larger it will blat
a lot more.

Are CW interferers really a problem? (not just sprogs from
the receiver!) I was going to make some sarcastic comment
about morse but I wont as last time I mentioned it got deluged
with email advertising the merits of morse code.

> --Johan
>
Nice talking to you Johan. Where on this small globe are you?
I can't work it out.

Bi Bi
- --
A.N. Martin   anthony.martin@gecm.com
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

35. Re: [HFSIG:2155] Re: Serial versus parallel, mainly parallel

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 20 Jun 1997 22:58:47 +0100
To: hfsig@tapr.org
Message-ID: <33AAFD17.358@dircon.co.uk>
Rendered body type: text/plain

Antony Martin wrote:
>
> Are CW interferers really a problem? (not just sprogs from
> the receiver!) I was going to make some sarcastic comment
> about morse but I wont as last time I mentioned it got deluged
> with email advertising the merits of morse code.

Hello Antony,

Yes they are, to name a few.

 Morse,
 Broadcast carriers,
 2 Tone FSK (FEK) signals

KV90 will find this out !

- CUL

Charles G4GUO.

36. Re: [HFSIG:2155] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 20 Jun 1997 16:06:34 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199706202306.QAA21728@servo.qualcomm.com>
Rendered body type: text/plain

I seem to recall writing a fairly long missive about raised-cosine
filtering on this list some months back. You can probably find it in the
archives.

To summarize, there are two requirements for optimum receive and
transmit filters on an AWGN channel:

1. The receive and transmit filters must be matched. This assumes that
the data ahead of the transmit filter consists of impulses, which have
a flat frequency spectrum. If not, e.g., if the raw transmit data are
square pulses, this represents an additional stage of filtering that

must be taken into account, usually by adding a correction to the
transmit filter.

2. The product of the transmit and receive filters (i.e., the overall
channel filter response) must have a frequency response that is
antisymmetric around the Nyquist frequency of one half of the data
rate.

One function that meets criterion #2 is the raised cosine. There are
others, but this one is easily implemented. To meet criterion #1,
simply take the square root of whatever function you've chosen and
make that the response of each filter. That makes the filters matched.
And the product of the square root response times itself gives the
original function.

Phil

37. PSK experiments.

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 20 Jun 1997 17:19:44 -0800
To: HFSIG@tapr.org
Message-ID: <199706210019.RAA15278@peak.org>
Rendered body type: text/plain

Hi folks,

I have been following the progress of a number of amateurs in the UK and
Europe on the development of a new PSK modem for HF. This modem runs at 31
BPS en uses some novel encoding - this is probably the closest thing to
replacing RTTY for round-table QSO's, except that it is so much more robust
and noise tolerant.

The guys on the other side of the Atlantic are listening for my beacon on
14070 USB which runs during the hours 2000 - 0200 UTC. Listen for the PSK
carrier and my CW-id.

If anyone else is interested in participating; all you need is a 56K EVM, a
PC, and tranceiver. Let me know and I'll be happy to provide the software to
run it - the software is really neat and has a nice-looking "cockpit." This
code was written by Peter Martinez, G3PLX, and modified by Izak, PA0IHD to
run a beacon.

Let me know if you hear anything.

73

--Johan, KC7WW

38. Re: Serial versus parallel, mainly parallel

From: "Bergzen Hakan, KR" <hakan.bergzen@enator.se>
Date: Sat, 21 Jun 1997 18:29:00 +0100
To: hfsig <hfsig@tapr.org>
Message-ID: <33B18DD5@noak>
Rendered body type: text/plain

Hello everybody,

I generally agree with the pro's and con's of parallel versus serial tone
modems as stated by Antony. However, on-air tests being conducted by several
different
organisations clearly shows a BER (or link availablility @ some BER
threshold)
advantage when using serial tone modems (on non-FH channels). The reasons
seems to depend on the embedded narrowband interference cancellation
function, the adaptive channel equaliser, and a good combination/interaction
of FEC and modulation format resulting in a very robust operation under
varying propagation conditions.

The ability to cancel one or more narrowband interferers within the 3 kHz
signal BW can also be viewed as an ability of finding many more usable
channels channels (i.e. those having a 3 kHz noise/interference level below
a certain threshold), which can be exploited by frequency adaptive systems.

The adaptive equaliser is necessary when the symbol time decreases. It would
be interesting to know whether it would also have an improving effect on
long symbol time waveforms (like parallel tone). One feature of the serial
tone modems is their diurnal robustness, whereas parallel tone modems have a
tendency of performing poorly during night-time. Perhaps an adaptive
equaliser could limit this effect? But then, is it at all possible to

extract multipath information from the parallel tone demodulator (as needed by the equaliser)? I guess a parallel tone modem can be viewed as a number of independently operating narrowband serial tone modems, so maybe it will be possible to combine side information from them all to come up with some multipath data for the full signal bandwidth.

Down-under, Stephen Cook/DSTO has shown promising results from Trellis Coded parallel tone modems (even though some argue that the results depend on the large interleaver depth used). Turbo codes will probably work even better (although there seems to be some licensing requirements if using that technique). Maybe more thought need to go into finding FEC's optimally suited for the number of parallel tones, and the type of m-PSK per tone?

Everybody today seems very focused on high-speed modems (in contrast to very robust modems). It is just as if you were to construct a 9600 bps HF modem that in real operation would have an effective ARQ throughput of only 200 bps, it would still sell better than a (comparatively cheap and) robust 300 bps M-ary FSK modem (or chirp or some other robust technique). As I see it, you only need as much bps power as your traffic load requires, thus if you are only sending store-and-forward messages (like email) then the end-user will not care if you actually only used the embedded 62.5 bps DTM mode in the 8-ary FSK MS-188-141A/FS-1045A ALE. So why not take advantage of one of the benefits of HF operation, namely the zero-per-minute cost ;-). The only time when there seem to be a real need for high-speed is on direct end-to-end links like digitised voice (excluding specific military applications).

By the way, the Nordic HF homepage will be awakened shortly (in time for the IEE Nottingham conference). I will post the web site address as soon as it is official.

Well, that is only my 2 cents of amateurish views.
Hakan (hakan.bergzen@enator.se)

39. Re: [HFSIG:2159] Re: Serial versus parallel, mainly parallel

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sun, 22 Jun 1997 01:53:45 -0400
To: hfsig@tapr.org
Message-ID: <33ACBDE9.379C6A3B@worldnet.att.net>
Rendered body type: text/plain

Bergzen Hakan, KR wrote:

> Hello everybody,
>
> I generally agree with the pro's and con's of parallel versus serial
> tone
> modems as stated by Antony. However, on-air tests being conducted by
> several
> different
> organisations clearly shows a BER (or link availablility @ some BER
> threshold)
> advantage when using serial tone modems (on non-FH channels). The
> reasons
> seems to depend on the embedded narrowband interference cancellation
> function, the adaptive channel equaliser, and a good
> combination/interaction
> of FEC and modulation format resulting in a very robust operation
> under
> varying propagation conditions.
>
>

I agree with everything Hakan has said. The advantages/disadvantages of

parallel vice serial are ones that need careful examination. Cook is
one of the
best authors on the subject and he will be a key note speaker at the HF
conference
in the UK in two weeks.

Some of the advantages of the parallel modem are due almost entirely to
the
not having a need for an adaptive equalizer and in some cases, (such as
CODAN)
no interleaver, etc. Yes the BER of serial modems in all cases I KNOW
of
are better but the cost is a very complex demodulator, one that took me
almost

6 months to completely master (that is in working code)

Bob

--
****

* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd. * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520 * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963 * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061 * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240 * Council. Brownie Troop 193
*
****

40. Stephen Cook's Longfish technical paper

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sun, 22 Jun 1997 03:33:42 -0400
To: hfsig@tapr.org
Message-ID: <33ACD556.F5F234A9@worldnet.att.net>
Rendered body type: text/plain

With all of the recent talk about parallel modems and someone mentioning
Cook's
paper, here is the CORRECT link, not the one incorrectly given by a
message in
the archives

http://www.itr.unisa.edu.au/~dstowww/longfish/hf95_6.html

Bob
--
****

* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd. * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520 * Post 995, Troop 5700 Advancement Chair,
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*
* (W) 609-279-6240 * Council. Brownie Troop 193
*
****

41. Re: [HFSIG:2161] Stephen Cook's Longfish technical paper

From: forrerj@peak.org (Johan Forrer)
Date: Sun, 22 Jun 1997 12:19:15 -0800
To: hfsig@tapr.org
Message-ID: <199706221918.MAA07632@peak.org>
Rendered body type: text/plain

Bob,

Your mail must be very late .... several folks have indicated the correct URL,
must have been weeks ago. Or are you only catching up now??

--Johan

>With all of the recent talk about parallel modems and someone mentioning
>Cook's
>paper, here is the CORRECT link, not the one incorrectly given by a
>message in
>the archives
>
>http://www.itr.unisa.edu.au/~dstowww/longfish/hf95_6.html
>
>Bob

```
>--
>*****************************************************************************
>
>* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
>Astronomy*
>* 64 Brooktree Rd.         * Amateur Radio, and Golf. Scouts: Advisor
>*
>* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
>*
>* (H) 609-443-8963         * Sanhican #2 (Brotherhood), Used to be a
>*
>* (F) 609-924-3061         * Buffalo (NE-III-120).  George Washington
>*
>* (W) 609-279-6240         * Council.  Brownie Troop 193
>*
>*****************************************************************************
>
>
>
>
```

42. 31 Baud PSK

```
From: forrerj@peak.org (Johan Forrer)
Date: Sun, 22 Jun 1997 14:53:22 -0800
To: HFSIG@tapr.org
Message-ID: <199706222152.OAA20795@peak.org>
Rendered body type: text/plain
```

Hi all,

I have had a lot of e-mail from kind folks
sending in signal reports of my beacon on 14070,
but also interest in participating in 31 BPSK.

I'll work at seeing that a code package and brief
description be made available for downloading
shortly. Please watch HFSIG for an announcement.

The waveform and protocol is intended for weak signal work,
thus the relatively long symbol time and narrow bandwidth
characteristics. The folks in Europe are actually monitoring
for a signal that probably is barely audible, and thus
require that we all are on the same frequency, at least
within a few Herz or so. The user interface has a couple of
aids to help you get there once you are within 10 Hz or so.

till later, 73

--Johan

43. Re: [HFSIG:2160] Re: Serial versus parallel, mainly parallel

```
From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Sun, 22 Jun 1997 18:34:14 -0500 (CDT)
To: hfsig@tapr.org
Cc: hq@arrl.org, wb5jbp@arrl.org, kj5ae@arrl.org
Message-ID: <Pine.SOL.3.91.970622182337.16824E-100000@www.kelly-afb.org>
Rendered body type: text/plain
```

We (U.S. hams) don't care which and my guess is that the ham around the
world in general doens't care either...all we want is a 1200 BPS robust
modem for HF that we can use with a DSP platform.

Give us something not just words....we're desperate -- Walt/K5YFW

```
=============================================================================
|        If I wanted "Windows", I'd knock a hole in my wall.                |
=============================================================================
|                                  |                                        |
|                                  | The Micro$oft operating system         |
| Walt DuBose - K5YFW              | didn't get as bad as it is             |
| E-Mail k5yfw@www.kelly-afb.org   | overnight.  I has taken over 10        |
| Business Telephone: (210)925-1171| years of careful, calculated,          |
|     Home Telephone: (210)696-3196| development.                           |
|                                  |                                        |
=============================================================================
```

44. Re: [HFSIG:2164] Re: Serial versus parallel, mainly parallel

From: forrerj@peak.org (Johan Forrer)
Date: Sun, 22 Jun 1997 20:04:16 -0800
To: hfsig@tapr.org
Message-ID: <199706230303.UAA27313@peak.org>
Rendered body type: text/plain

Hi Walt,

>We (U.S. hams) don't care which and my guess is that the ham around the
>world in general doens't care either...all we want is a 1200 BPS robust
>modem for HF that we can use with a DSP platform.

If I may remind folks interested in a high speed modem with CSMA capabilities
that can show an impressive improvenet over existing 300 baud HF Packet:

Presently we have a parallel HF modem that is capable of 3000+ raw bps
(uncoded.)
The modem was demonstrated at the 1994 DCC venue in Texas and has since been
improved. There always is the possibility of make further improvements and
going
the extra mile, given that we see that the technology are put to some use
somewhere.

We have recently seen the reports from Finnish hams using this modem on
the 80M band, which is a tough one to deal with, so we know that it has
potential.
This modem runs on a 56K EVM ($150) and requires a PC running some flavor of
NOS
(even a lowly 386 will work) - this you will agree is not extravagent. It is
rather
unfortunate that we have not seen any takers here in the USA.

What can we do to help get US amateurs involved and interested in trying out
the system?
That, I think, is the main hurdle, but I would like to hear your opinion.

73,

--Johan

45. Re: [HFSIG:2164] Re: Serial versus parallel, mainly parallel

From: Bill Carver <bcarver@magiclink.com>
Date: Mon, 23 Jun 1997 00:20:20 -0500 (CDT)
To: hfsig@tapr.org, hfsig@tapr.org
Message-ID: <199706230520.AAA01537@tapr.org>
Rendered body type: text/plain

At 06:37 PM 6/22/97 -0500, Walt DuBose - K5YFW wrote:
>We (U.S. hams) don't care which and my guess is that the ham around the
>world in general doens't care either...all we want is a 1200 BPS robust
>modem for HF that we can use with a DSP platform.
>
>Give us something not just words....we're desperate -- Walt/K5YFW

I'm not desperate, just a little tired of words. I'll take 1200 BPS, but
would trade it for one-tenth that in a heartbeat if it had measureably
better error rate in presence of serious HF selective fading. But whatever,
WHENEVER, it needs to be available worldwide at a reasonable price so
worldwide digital folks can climb on the same bandwagon and talk to each other.

Bill - W7AAZ

46. Re: [HFSIG:2165] Re: Serial versus parallel, mainly parallel

From: Charles Brain <chbrain@dircon.co.uk>
Date: Mon, 23 Jun 1997 06:35:48 +0100
To: hfsig@tapr.org
Message-ID: <33AE0B34.13D7@dircon.co.uk>
Rendered body type: text/plain

Johan Forrer wrote:
>
> Presently we have a parallel HF modem that is capable of 3000+ raw bps
> (uncoded.)
> unfortunate that we have not seen any takers here in the USA.
>
> What can we do to help get US amateurs involved and interested in trying out
> the system?

> That, I think, is the main hurdle, but I would like to hear your opinion.
>
> 73,
>
> --Johan

Does it conform to the US HF restrictions on High speed data waveforms ?
Just out of curiosity!

- Charles G4GUO

47. RE: [HFSIG:2153] Re: Serial versus parallel, mainly parallel

From: jalocha@axdele.cern.ch
Date: Mon, 23 Jun 1997 16:05:01 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199706231405.QAA18656@dxmint.cern.ch>
Rendered body type: text/plain

>Our experimental efforts are using m-DQPSK. Phil Karn (the famous KA9Q - Mr.
>TCP/IP) is suggesting the offset form (SQPSK).

Could someone please remind me shortly why the shifted form is better ?
I am building up a new parallel tone modem and I want to use
the SQPSK but I must say I don't know why should I use it ? :-)

If one gets a good idea for a good but simple FEC operating
on 60-64 bits blocks please let me know too.
16 (or 15 ?) RS 4-bit symbols ?

Thanks, Pawel.

48. RE: [HFSIG:2167] Re: Serial versus parallel, mainly parallel

From: jalocha@axdele.cern.ch
Date: Mon, 23 Jun 1997 16:11:03 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199706231411.QAA21812@dxmint.cern.ch>
Rendered body type: text/plain

>Does it conform to the US HF restrictions on High speed data waveforms ?
>Just out of curiosity!

While writing the code I was not aware of any restrictions
(even if there were any I would not care for these experiments)
so the modem may not not conform to them...
Could you please tell me (us) what they are in essence ?

Pawel

49. RE: [HFSIG:2165] Re: Serial versus parallel, mainly parallel

From: Victor Kean <vkean@web.compuserve.com>
Date: Mon, 23 Jun 1997 10:23:37 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=Web%l=CARMINA-970623142337Z-5668@carmina.web.compuserve.com>
Rendered body type: text/plain

I would be interested in trying this out. Where do I sign up? I
already have the EVM 56K and appropriate computers.

Vic, k1lt

>-----Original Message-----
>From: forrerj@peak.org [SMTP:forrerj@peak.org]
>Sent: Sunday, June 22, 1997 11:08 PM
>To: hfsig@tapr.org
>Subject: [HFSIG:2165] Re: Serial versus parallel, mainly parallel
>
>[Victor Kean] <stuff deleted>
>
>What can we do to help get US amateurs involved and interested in trying out
>the system?
>That, I think, is the main hurdle, but I would like to hear your opinion.
>
>
>73,
>
>--Johan
>

50. RE: [HFSIG:2167] Re: Serial versus parallel, mainly parallel

From: jalocha@axdele.cern.ch
Date: Mon, 23 Jun 1997 16:31:46 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199706231431.QAA24796@dxmint.cern.ch>
Rendered body type: text/plain

A "what if" kind of question:

What I take the 30-tone, 50 baud QPSK modem and scale it down by say
a factor of 8 so I get baudrate of 6-7/s, total bandwidth around 300 Hz
and total bit rate like 300 bps.
Would HF support this low baud rate (this would be phase modulation).
I remember somebody saying that on HF the phases are consistant
up to about 1/10th of a second ?

This came to my mind because the SLOWBPSK was made by scaling down
the satellite 1200 bps BPSK modem, so why not scale the "fast" multi-tone
modem and get a "slow". People asking for low bandwidth (not bigger than
RTTY), going "wide band" (several tones + FEC), low symbol rate
(even lower than for the SLOWBPSK) would be satisfied ?

Pawel

51. Re: [HFSIG:2167] Re: Serial versus parallel, mainly parallel

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 23 Jun 1997 08:32:14 -0800
To: hfsig@tapr.org
Message-ID: <199706231531.IAA00726@peak.org>
Rendered body type: text/plain

Hello Charles,

>Does it conform to the US HF restrictions on High speed data waveforms ?
>Just out of curiosity!
>
>- Charles G4GUO
>

We made some effort to ensure that it is.

1) After the 1994 DCC, Paul Rinaldo, the legal folks at ARRL, checked with
the FCC in Washinton whether the use of an SSB-bandwidth for digital was
legan; that was cleared OK. The message was that it would be hated, and will
be opposed by others sharing the band segment, but it was legal, nevertheless.

2) The second issue is whether the protocol is publicly plublished. That is
a little more involved, but in my opinion is OK. First, the source code is
in public domain, although the author's copyright states that the code not
be used for commercial or military applications. Second, HFSIG archives
contains full disclosure of how it works, probably not in a nice formal
document, but it is there. I also wrote brief description of the protocol
and specifications for HFSIG in PSR some time ago. Perhaps it is time for a
good article on it. Perhaps Pawel, SP9VRC, the author, may be interested?

Did I leave anything out?

--Johan

52. Re: [HFSIG:2168] Re: Serial versus parallel, mainly parallel

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 23 Jun 1997 08:35:31 -0800
To: hfsig@tapr.org
Message-ID: <199706231535.IAA01135@peak.org>
Rendered body type: text/plain

Pawel,

>>Our experimental efforts are using m-DQPSK. Phil Karn (the famous KA9Q - Mr.
>>TCP/IP) is suggesting the offset form (SQPSK).
>
>Could someone please remind me shortly why the shifted form is better ?
>I am building up a new parallel tone modem and I want to use
>the SQPSK but I must say I don't know why should I use it ? :-)

It yields a constant signal envelope as far as I can remember, but that has
some quirks that confounds carrier/clock recovery. Perhaps Phil can provide

a better description.

--Johan

53. Re: [HFSIG:2169] Re: Serial versus parallel, mainly parallel

From: Charles Brain <chbrain@dircon.co.uk>
Date: Mon, 23 Jun 1997 17:02:20 +0100
To: hfsig@tapr.org
Message-ID: <33AE9E0C.5873@dircon.co.uk>
Rendered body type: text/plain

jalocha@axdele.cern.ch wrote:
>
> >Does it conform to the US HF restrictions on High speed data waveforms ?
> >Just out of curiosity!
>
> While writing the code I was not aware of any restrictions
> (even if there were any I would not care for these experiments)
> so the modem may not not conform to them...
> Could you please tell me (us) what they are in essence ?
>
> Pawel

I have been told not to worry about this Pawel by private email.
I don't have a copy of the US regs.

I just seemed to remember a discussion on the sig about this last year.

- Charles

54. Re: [HFSIG:2171] Re: Serial versus parallel, mainly parallel

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 23 Jun 1997 09:18:16 -0800
To: hfsig@tapr.org
Message-ID: <199706231617.JAA08652@peak.org>
Rendered body type: text/plain

Hello Pawel,

My 2 cents worth ....

>A "what if" kind of question:
>
>What I take the 30-tone, 50 baud QPSK modem and scale it down by say
>a factor of 8 so I get baudrate of 6-7/s, total bandwidth around 300 Hz
>and total bit rate like 300 bps.
>Would HF support this low baud rate (this would be phase modulation).
>I remember somebody saying that on HF the phases are consistant
>up to about 1/10th of a second ?

Yes, technically this may work from the impression that I get from those
working at parallel HF modems; that you do not need equalization at low baud
rates (I assume you are scaling the baudrate down 50/8=6.25).
Are you thinking about this because of the narrow bandwidth or from a
robustness point of view?

A bit of rambling:
In my opinion, there is a need for several digital modes that I think will
excite people - just a few ideas here:

A) Keyboard to keyboard conversations (I) (for good signals:) As good or
better than Pactor II for speed/robustness/interactiveness. Bandwidth approx
500 Hz. Maximum throughput roughly 900 bps.
It is evident that this will be some form of an ARQ system with low
operating overheads.

B) Keyboard to keyboard conversations (II) (for weak signal work:) Either
narrow BW or some form of spreading.
Use innovative methods to gain the best in coding gains possible.

C) Networking. At least 1200 bps robust throughput. SSB bandwidth, good ECC.
Evidently some form of CSMA.

So, you want to sell your idea to a bunch of amateurs, where would you like
to see it fit in above scheme?

My thoughts, comments?

--Johan

55. Re: [HFSIG:2170] Re: Serial versus parallel, mainly parallel

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 23 Jun 1997 09:18:17 -0800
To: hfsig@tapr.org
Message-ID: <199706231617.JAA08656@peak.org>
Rendered body type: text/plain

Hello Vic,

>I would be interested in trying this out. Where do I sign up? I
>already have the EVM 56K and appropriate computers.
>
>Vic, k1lt

No signing up needed; The first thing would be to obtain the latest version
of NEWQPSK. This is available from the TAPR DSP "recent uploads" file area.
I placed the most recent versions of Pawel's 1200 AX.25 modem and the
multitone HF modem there.

If you have not already built up an interface, it is a good time to do that.
Remember you will be running the SSB tranceiver in sideband mode, and you
need to control PTT, probably also frequency UP/DOWN, however, the latter is
not essential.

If you are not very familiar with using the EVM, take a bit of time to get
to know it. Unfortunatly we still don't have turnkey recipes for everything.
Ask if you need help - there are talented folks monitoring the list that are
eager to help.

Let us know how it goes, I am very interested in running some tests with you.

Good luck, 73

--Johan

56. Re: [HFSIG:2172] Re: Serial versus parallel, mainly parallel

From: Charles Brain <chbrain@dircon.co.uk>
Date: Mon, 23 Jun 1997 17:34:22 +0100
To: hfsig@tapr.org
Message-ID: <33AEA58E.7B01@dircon.co.uk>
Rendered body type: text/plain

Johan Forrer wrote:
>
 1) After the 1994 DCC, Paul Rinaldo, the legal folks at ARRL, checked
with
> the FCC in Washinton whether the use of an SSB-bandwidth for digital was
> legan; that was cleared OK. The message was that it would be hated, and will
> be opposed by others sharing the band segment, but it was legal, nevertheless.
>
>
> Did I leave anything out?
>
> --Johan

No thats fine,

I remember there was some discussion on this last year.

I think the bit about other amateurs objecting is very true. However if
you
stick to shared bands then no-one is going to know that you are not some
military user!

- Charles.

57. FFT, Integrate & dump, matched filters

From: mcdermot@rockdal.aud.alcatel.com (Tom Mcdermott)
Date: Mon, 23 Jun 97 15:47:14 CDT
To: hfsig@tapr.org
Message-ID: <9706232047.AA03862@eagle.aud.alcatel.com>
Rendered body type: text/plain

In response to some questions and comments, here is short version
of matched filters:

1. Integrate and dump filter is matched only to 'rectangular'
   transmit data.

2. Root-raised-cosine is a good choice for many applications.
   RRC in Tx and same RRC in receiver giving overall RC response
   plus channel distortion.

3. An FFT is equivalent to a channel filter followed by an
   integrate-and-dump. This is sort of obvious when you think
   about how the FFT works. So it is best only for rectangular
   data pulses.

- Tom

Tom McDermott
mcdermot@aud.alcatel.com

58. Re: [HFSIG:2178] FFT, Integrate & dump, matched filters

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 23 Jun 1997 15:44:57 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199706232244.PAA07080@servo.qualcomm.com>
Rendered body type: text/plain

>1. Integrate and dump filter is matched only to 'rectangular'
>   transmit data.

True unless the data is windowed by a local copy of the transmitted
pulse shape. No window is actually equivalent to a rectangular window.

>3. An FFT is equivalent to a channel filter followed by an
>   integrate-and-dump. This is sort of obvious when you think
>   about how the FFT works. So it is best only for rectangular
>   data pulses.

Again, true unless the raw received data is first windowed by the pulse
shape.

As an aside, since multiplication in the time domain (e.g., by a
window function) is equivalent to convolution in the frequency domain,
the effect of the usual time-domain window is to "low pass
filter" the frequency domain information. (More accurately, the
resulting spectrum is the convolution of the spectra of the original
signal and the impulse response of the window function, which is
presumed to resemble a low pass filter.)

This is why the usual non-rectangular windowing functions (e.g.,
Hamming, Hanning, Blackman, Kaiser) have the effect of broadening the
main lobe of the frequency spectrum and of knocking down the amplitude
of the (rapidly varying) sidelobes.

Frequency-domain convolution is generally not done with a FFT because
it's much easier to just do the time-domain multiplication. But the
reverse is true on a conventional analog spectrum analyzer. The "video
bandwidth" setting is equivalent to a time-domain windowing function.

Phil

59. Re: [HFSIG:2175] Re: Serial versus parallel, mainly parallel

From: Bill Carver <bcarver@magiclink.com>
Date: Mon, 23 Jun 1997 19:34:55 -0500 (CDT)
To: hfsig@tapr.org, hfsig@tapr.org
Message-ID: <199706240034.TAA02386@tapr.org>
Rendered body type: text/plain

At 11:23 AM 6/23/97 -0500, Johan Forrer wrote:
>A bit of rambling:
>In my opinion, there is a need for several digital modes that I think will
>excite people - just a few ideas here:
>
>A) Keyboard to keyboard conversations (I) (for good signals:) As good or
>better than Pactor II for speed/robustness/interactiveness

>B) Keyboard to keyboard conversations (II) (for weak signal work:) Either

>narrow BW or some form of spreading.
>Use innovative methods to gain the best in coding gains possible.
>
>C) Networking. At least 1200 bps robust throughput. SSB bandwidth, good ECC.
>Evidently some form of CSMA.

>So, you want to sell your idea to a bunch of amateurs, where would you like
>to see it fit in above scheme?

#1 - B (which includes A: except for speed)
#2 - A

C: doesn't fit in anywhere in my opinion. Locally our packet nodes are
being moved down from 10,000+ ft locations to reduce coverage, and the
bulletin board function is being deleted from more than one site.

Fact: bulletin boards fill up with forwarded nonsense and mostly junk,
consuming time and wasting spectrum. Automated generation and forwarding
squeezes out other forms of communcation.

Bill - W7AAZ

60. Re: [HFSIG:2180] Re: Serial versus parallel, mainly parallel

From: Lyle Johnson <lylej@azstarnet.com>
Date: Mon, 23 Jun 1997 18:48:52 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19970623184847.0076ef20@pop.azstarnet.com>
Rendered body type: text/plain

>Fact: bulletin boards fill up with forwarded nonsense and mostly junk,
>consuming time and wasting spectrum. Automated generation and forwarding
>squeezes out other forms of communcation.

This applies to broadcast TV as well, but some people actually watch it and
claim to enjoy it...

If we can more efficiently use HF as part of networking, we should (since
we're already using it for this urpose). Content/judgment is left to the
users.

Cheers,

Lyle

61. Robust fast HF modem

From: sivula@NCSNRS01ES.ntc.nokia.com (Sivula Timo NTC/Espoo)
Date: Tue, 24 Jun 1997 05:07:52 +0300
To: hfsig@tapr.org (hfsig)
Message-ID: <1997Jun24.042936.1604.2261091@ntcit-mmta6.ntc.nokia.com>
Rendered body type: text/plain

Hello there

>We have recently seen the reports from Finnish hams using this modem on
>the 80M band, which is a tough one to deal with, so we know that it has
>potential.
>This modem runs on a 56K EVM ($150) and requires a PC running some flavor
of
>NOS
>(even a lowly 386 will work) - this you will agree is not extravagent. It
is

Well, In fact a 286 will do, you only need a packet radio terminal that
supports KISS. There are packer terminals that work on 8088 processors as
well that support KISS. I think you find these XT's anywhere for virtually
free nowadays.

The conclusions we have from our tests here is Finland are very promising,
we are runnig FBB forward over this modem for some time already on 80m. It
works quite ok.

If somebody is interested in having a sked to Finland using this new modem,
I would be interested in being at the european end.

Timo Sivula, OH6KK & OH2LVZ
Nokia Telecommunications
New Radio Systems Marketing & Business Development
mobile: +358 (0)40 5024027 office: +358 (0)9 511 38426

62. unsubscribe

From: Rick Contaldo <rickc@tridelta.com>
Date: Tue, 24 Jun 1997 08:00:42
To: hfsig@tapr.org
Message-ID: <3.0.1.16.19970624080042.234fe4ee@pophost.tridelta.com>
Rendered body type: text/plain

[Empty message body]

63. Re: Serial versus parallel

From: "Bergzen Hakan, KR" <hakan.bergzen@enator.se>
Date: Tue, 24 Jun 1997 15:03:00 +0100
To: hfsig <hfsig@tapr.org>
Message-ID: <33B551EC@noak>
Rendered body type: text/plain

Antony Martin wrote:
>I'm very suprised that data directed equiliser updating is patented;
>when was this patented? ...snip...

In my version of the Harris HF modem RF-5710 broschure there is a reference
to
a US Patent no 4,365,338 which at http://www.uspto.gov/
is said to be a Harris patent entitled "Technique for high rate digital
transmission over a dynamic dispersive channel". Apparently it was patented
1980. A full description can be ordered from that web site.

The US patent office (see site above) will generate quite a lot of patents
on
search words such as (equaliser or equalizer).

/Hakan

64. Participation of HF PSK tests in USA

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 24 Jun 1997 09:17:17 -0800
To: DSP@tapr.org, HFSIG@tapr.org
Message-ID: <199706241616.JAA05465@peak.org>
Rendered body type: text/plain

Hi folks,

I would very much like do some local PSK testing. Is there anyone that
perhaps are set up to run some tests, preferably on the west coast, but
elsewere in US would be OK too.

Perhaps we can set aside a day/time for a round table QSO using the existing
31 baud PSK and also the parallel tone system.

Please e-mail me directly - I'll post a notice when we get something arranged.

Thanks much for your interest.

Johan, KC7WW

65. m-FSK Viterbi-based clock extraction

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 24 Jun 1997 10:37:17 -0800
To: HFSIG@tapr.org
Message-ID: <199706241736.KAA15649@peak.org>
Rendered body type: text/plain

Hello,

I would like to try my hand at code-assisted bit extraction (CABS) on a
m-FSK scheme and invite some discussion:

The idea behind CABS is to build the decoding trellis at sample rate, not at
symbol rate, with the intent to find the best instance, i.e., which sample
point index corresponds to the bit clock. It uses the Viterbi algorithm
(dynamic programming) to do this ofcourse.

Thinking aloud now ....

Say we have 8-FSK (one out of eight tones) - produced by a FFT. What we
would like to achieve is to extract a bit clock from the demodulator without
looking for zero crossings or bit edges, because that does not work too well
with low, noisy signal levels. We also do NOT want to use things like DPPL's

because that also requires substantial S/N to work decently.

A good indicator for the bit clock is to use some non-linear device that
uses the FFT outputs to sum up pulses from the FFT bins. This would be great
if we are assured that there always are transitions ocurring between tones.
When there are successive periods of the same tone, this scheme does not
work. One may advocate the use of scramblers to force transitions, but how
do we do this on existing schemes where we cannot change the modulator in
the first place?

So, this is where the soft-decision Viterbi decoder comes in ....
At each sample instance, we produce value (a soft-decision value) from the
FFT array, telling the confidence we have about that measurement (several
suggestions were put forward on how to do this.) Further, lets assume that
the sample rate is 16 times the bit rate, thus there will be one instance
out of 16 that matches the correct bit sample time.

With this soft-valued data stream at hand, we now want to build the decoder
tellis to find the appropriate bit sample point. To use the Viterbi
algorithm, we need to define a suitable metric. I'm not to sure how to do
this, but one method that I thought of trying, was to use "proto-waves" (aka
G3RUH) as references. That then will produce 16 parallel trelisses each
producing its version of a winning path. Of the 16 trellises one (hopefully)
would produce a winning path with the highest confidence - this is the one
that tells us the right bit clock instance.

Are there any better ideas for the metric? I have a feeling that there might be.

Suggestions of further ideas would be appreciated.

Thanks much in advance,

--Johan

66. hsp50016 ddc

From: Stephen PEARCE <spearce@ccu1.auckland.ac.nz>
Date: Wed, 25 Jun 1997 13:24:57 +1200 (NZT)
To: hfsig@tapr.org
Message-ID: <199706250124.NAA07576@ccu1.auckland.ac.nz>
Rendered body type: text/plain

Hi,..
I am trying to obtain this device to experiment with the construction of an
HF receiver along the lines of the qex articles which appeared over the last 3
-4 years. Does anyone know where I could purchase it in 1 off quantities or
samples .. I have had no luck with the agents here in NZ and Harris have not replied.

Also looking for one offs of 10 or 12 bit adcs suitable to go with it
(such as ad9070 100 MSPS 10 bit adc or Burr Brown ads800 12 bit 40 MSPS)

Any assistance or advice gratefully accepted
Thanks
Stephen Pearce zl1any

67. Re: [HFSIG:2173] Re: Serial versus parallel, mainly parallel

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Wed, 25 Jun 1997 10:09:28 +0200
To: hfsig@tapr.org
Message-ID: <33B0D238.3F90D3B0@zeus.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

Johan Forrer wrote:
>
> Pawel,
>
> >>Our experimental efforts are using m-DQPSK. Phil Karn (the famous KA9Q - Mr.
> >>TCP/IP) is suggesting the offset form (SQPSK).
> >
> >Could someone please remind me shortly why the shifted form is better ?
> >I am building up a new parallel tone modem and I want to use
> >the SQPSK but I must say I don't know why should I use it ? :-)
>
> It yields a constant signal envelope as far as I can remember, but that has
> some quirks that confounds carrier/clock recovery. Perhaps Phil can provide
> a better description.
>
> --Johan

Constant envelope on each carrier of an OFDM transmission is useless as
the overall signal is of course not constant envelope. So this can't be
the reason for using SQPSK. Furthemore SQPSK (or OQPSK) can't be easily
demodulated non-coherently.

If you want to try something "different" try DAPSK, I've done some
simulations that show that 16-DAPSK is possible on HF if the S/N is good
enough. DAPSK is a combination of differential PSK and ASK, that can be
demodulated non-coherently.

73' Alexander

--
------------------------------+--------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| technik u. Elektroakustik | Fax : +49-6151-165545
|
| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de
|
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
------------------------------+--------------------------------------

68. Re: [HFSIG:2184] Re: Serial versus parallel

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Wed, 25 Jun 1997 10:23:14 +0200
To: hfsig@tapr.org
Message-ID: <33B0D572.E37CF02@zeus.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

Bergzen Hakan, KR wrote:
>
> Antony Martin wrote:
> >I'm very suprised that data directed equiliser updating is patented;
> >when was this patented? ...snip...
>
> In my version of the Harris HF modem RF-5710 broschure there is a reference
> to
> a US Patent no 4,365,338 which at http://www.uspto.gov/
> is said to be a Harris patent entitled "Technique for high rate digital
> transmission over a dynamic dispersive channel". Apparently it was patented
> 1980. A full description can be ordered from that web site.
>
> The US patent office (see site above) will generate quite a lot of patents
> on
> search words such as (equaliser or equalizer).
>
> /Hakan

You can download a copy from the IBM patent server

http://patent.womplex.ibm.com/details?patent_number=4365338

73' Alexander
--
------------------------------+--------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| technik u. Elektroakustik | Fax : +49-6151-165545
|
| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de
|
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
------------------------------+--------------------------------------

69. Microstrip antenna design - help !

From: Ted Manolescu <tmanolescu@instantel.com>
Date: Wed, 25 Jun 1997 12:15:42 -0400
To: "'ss@tapr.org'" <ss@tapr.org>
Cc: "'hfsig@tapr.org'" <hfsig@tapr.org>, "'ham@sss.org'" <ham@sss.org>
Message-ID: <13684FD9F298D011AD8800609769CE77071247@mailhost.Instantel.com>

Rendered body type: text/plain

Hi everybody !
  Does anybody know some references (technical literature and software
tools design) for microstrip antenna ?
  Thanks in advance !
  Ted Manolescu
  613 592 4642

70. Re: [HFSIG:2187] hsp50016 ddc

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 25 Jun 1997 21:08:19 -0800
To: hfsig@tapr.org
Message-ID: <199706260407.VAA20432@peak.org>
Rendered body type: text/plain

Stephan,

Yes - Farnell here in the USA will sell them in single quantities. I do not
have their contact info ready, perhaps you can follow up on this.

73

-Johan, KC7WW

>Hi,..
>   I am trying to obtain this device to experiment with the construction of an
>HF receiver along the lines of the qex articles which appeared over the last 3
>-4 years. Does anyone know where I could purchase it in 1 off quantities or
>samples .. I have had no luck with the agents here in NZ and Harris have
not replied.
>
>Also looking for one offs of 10 or 12 bit adcs suitable to go with it
>(such as ad9070 100 MSPS 10 bit adc or Burr Brown ads800 12 bit 40 MSPS)
>
>   Any assistance or advice gratefully accepted
>   Thanks
>   Stephen Pearce zl1any
>
>
>

71. Re: [HFSIG:2189] Re: Serial versus parallel

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Thu, 26 Jun 1997 00:27:03 -0400
To: hfsig@tapr.org
Message-ID: <33B1EF97.D50DFB35@worldnet.att.net>
Rendered body type: text/plain

Alexander Kurpiers wrote:

> Bergzen Hakan, KR wrote:
>
> You can download a copy from the IBM patent server
>
> http://patent.womplex.ibm.com/details?patent_number=4365338
>
> 73' Alexander
> --

Thanks. It doesn't contain the figures, but for our purposes, I think
the patent has expired. It was
filed 17 years ago.

--
****

* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193

*
****

72. Blaster TeLetype V 1.5 Released

From: Robert.Glassey@nmp.nokia.com
Date: Thu, 26 Jun 97 09:05:53 +0100
To: BitBucket@qsl.net, hfsig@tapr.org, ajr@ari.net, Andy.Wright@gpsemi.com,
bobg@epix.net, castellano@arrakis.es, darrellm@drmoody.demon.co.uk,
dhabecker@worldnet.att.net, dwinn@tng.net, forrerj@peak.org,
gc@bnl.gov, ik2ley.fabio@iol.it, jalarson@micron.com, jayt@iea.com,
jcullum@micron.com, juhola@ibm.net, lance@digital.net,
Magnus.Hammarstedt@midnet.telia.com, MN@NTCSPS01HE.ntc.nokia.com,
mto@nic.funet.fi, N.Campbell@ping.be, no2tj@mail.cybernex.net,
oh2gi@pcuf.fi, ON4VT@ping.be, rafael@globalmail.net,
robert.tyrakowski@lauterbach.com, ross@hypertools.com,
rstudio@ns.enet.cz, s51ay@s55tcp.ampr.org, SCHROEDERE@thmulti.com,
Sickingers@aol.com, sswhite@mcleod.net, steve@myna.com,
thopps@accessnt.com.au, w9hmp@juno.com, wa2mzf@juno.com,
wa2wim@ismi.net, whitener@hiwaay.net
Message-ID: <H0000292043c92a2@MHS>
Rendered body type: text/plain

Hi Everyone!!

```
***************    A N N O U N C I N G    ***************

  ***    *****    *             *   *     *       *****
  *  *     *      *             *   *     *       *
  ***      *      *              * *      *       ****
  *  *     *      *              * *      *           *
  ***      *      ****             *      *   *   ****
```

****

Yes, BTL V1.5 is finally available via the WWW from my homepage:

http://www.geocities.com/SiliconValley/Heights/4477/

This is quite close to what was originally promised for V2.0,
and is well beyond what I thought V1.5 would be.

Since V1.33 Beta there have been a few new features added, and a few
bugs fixed.

o End Of File charactors in Brag and Function Key buffers.
o CTRL-HOME clears function key buffer when editing.
o Output level is adjustable.
o Up/Down Arrow keys select feilds on Sound Blaster Control Panel.
o HELP updated, spell checked, and corrected.
o BUGFIX: W indicator doesn't vanish when nothing else is showing.
o BUGFIX: Baud rate is actually set on startup, not just displayed!

I hope you all enjoy using BTL as much as I have enjoyed
writing it. Many, many thanks to all those folks out there
who have tried BTL and given me feedback. Your tests, comments
and enthusiastic support for BTL has made it what it is today!
I hope BTL is now a stable, robust, widely compatible and
user friendly program that will encourage many many people
to take up digital modes and experiance the thrill of operating
a new mode. Now any radio amateur with a PC, sound card and
transciever can be on air with RTTY in minutes, for FREE!
(BTL is great for listeners too!)

E N J O Y ! ! !

Cheers,

Rob, OH2/G0VTQ

73. Re: [HFSIG:2189] Modem patents & reports

From: Antony Martin <anthony.martin@gecm.com>
Date: Thu, 26 Jun 1997 16:54:32 +0100
To: hfsig@tapr.org
Message-ID: <970626165432.13955@oas01.gmcl.co.uk>
Rendered body type: text/plain

> Bergzen Hakan, KR wrote:
> >

> > Antony Martin wrote:
> > >I'm very suprised that data directed equiliser updating is patented;
> > >when was this patented? ...snip...
> > >
> > In my version of the Harris HF modem RF-5710 broschure there is a reference
> > to
> > a US Patent no 4,365,338 which at http://www.uspto.gov/
> > is said to be a Harris patent entitled "Technique for high rate digital
> > transmission over a dynamic dispersive channel". Apparently it was patented
> > 1980. A full description can be ordered from that web site.
> > >
> > The US patent office (see site above) will generate quite a lot of patents
> > on
> > search words such as (equaliser or equalizer).
> > >
> > /Hakan
> >
> You can download a copy from the IBM patent server
>
> http://patent.womplex.ibm.com/details?patent_number=4365338
>
>
> 73' Alexander
> --

I've seen an abstract of the serial modem patent. Basically it says
that if you mix known bits with the traffic bits at the transmitter
you can use the known bits at the receiver to adapt the equaliser.

This is a bit of a bugger as you probably have to pay money to harris
if you want to make STANAG 4285 or MIL STD 188 110A modems or do
software for it. It's not as though the FEC in these standards is ideal,
they've just become accepted standards.

While browsing at the patent office I noticed a lot of other patents
regarding modulation/demodulation methods, DSP algorithms, and
frequency hopping. You'd be suprised what innocuous & obvious things
have been patented; such as digital downconversion using SIN & COS lookup
tables, and subsampling a receiver IF.

What exactly is the position of amateurs & patents? Is it OK to share
modem software which is covered by patents? I suppose it depends on
whether it's experimental, a prototype, or whether it's free or you're
selling it.

The DNS lookup seems to have died here so I'll look at
patent.womplex.ibm.com some other time.

*** If you haven't seen
http://www.itr.unisa.edu.au/~dstowww/longfish/hf95_6.html then
I suggest you do as it's quite a thorough report on modem performance
vs. channel conditions. There's a number of odd and unsubstantiated
statements in it but a lot of work has been done in producing it.

Having read it I might have an (experimental) go at quadrupling my
modems' tones to take it to 9600 raw bits/sec.

- --
A.N. Martin   anthony.martin@gecm.com
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

74. Re: [HFSIG:2149] Re: Serial versus parallel

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 26 Jun 1997 21:50:05 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199706270450.VAA27218@servo.qualcomm.com>
Rendered body type: text/plain

>Do you also advocate the use of guard bands in the OFDM demodulation
>process, i.e., instead of using all the data from the entire symbol
>time, the FFT process only uses a slice out of the middle of the baud,
>thus leaving a guard band each side to absorb multipath jitter?

Note this can be accomplished with a window function on each pulse.
You can use any of the traditional windowing functions: Hamming, Hanning,
Blackman, Kaiser, etc.

>Continuing on the issue of OFDM; Some of us have been playing around with
>RC pulse shaping applied to individual carriers within the OFDM ensamble,
>only to find that one then needs wider spacing the 1/T orthogonal spacing. The

Of course! This is exactly the same phenomenon as a windowing function
applied to the input of a FFT knocking down the sidelobe leakage at
the cost of broadening the main lobe. Pick your poison; you can't have
both a narrow main lobe and low sidelobes.

Phil

75. dsp56002 evm

From: Stephen PEARCE <spearce@ccu1.auckland.ac.nz>
Date: Tue, 1 Jul 1997 01:33:11 +1200 (NZT)
To: hfsig@tapr.org
Message-ID: <199706301333.BAA01041@ccu1.auckland.ac.nz>
Rendered body type: text/plain

Hi
        Now that the group purchase of these evm s is closed what is
the best way of obtaining one?

        Thanks
        Stephen pearce zl1any

76. Re: [HFSIG:2194] Re: Modem patents & reports

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Mon, 30 Jun 1997 09:04:06 -0500 (CDT)
To: hfsig@tapr.org
Cc: kjh@usc.edu, kjh@apollo.bv.net, debbie@texas.net, dreid@tis.kelly.af.mil
Message-ID: <Pine.SOL.3.91.970630084355.16214B-100000@www.kelly-afb.org>
Rendered body type: text/plain

On Sun, 29 Jun 1997, Antony Martin wrote:

> > Bergzen Hakan, KR wrote:
> This is a bit of a bugger as you probably have to pay money to harris
> if you want to make STANAG 4285 or MIL STD 188 110A modems or do
> software for it. It's not as though the FEC in these standards is ideal,
> they've just become accepted standards.

        I don't know about STANAG 4285 but MIL STD 188 110A and even MIL STD
        118 110C are DoD Standards and fully published. Several manufacturers
        make MIL STD 188 110C modems.

        I think the 110C Change Notice 2 is the Harris Serial Tone Modem
        and this probably is restricted.

        With all due respect to the Harris Manager who monitors this SIG,
        I've used both in an almost side-by-side operation and there is
        to my thoughts only a slight advantage seen using the serial tone
        modem vs a 39 parallel tone MIL STD modem. For ham radio use,
        I cannot see the advantage to using the serial tone modem if it is
        all that more complicated to build/program.

        Someone on the SIG has hard copies of MIL-STD-188-110C.

Walt/K5YFW

    ==
    | If I wanted "Windows", I'd knock a hole in my wall. |
    ==
    | | |
    | | The Micro$oft operating system |
    | Walt DuBose - K5YFW | didn't get as bad as it is |
    | E-Mail k5yfw@www.kelly-afb.org | overnight. I has taken over 10 |
    | Business Telephone: (210)925-1171 | years of careful, calculated, |
    | Home Telephone: (210)696-3196 | development. |
    | | |
    ==

77. RE: [HFSIG:2175] Re: Serial versus parallel, mainly parallel

From: jalocha@axdele.cern.ch
Date: Mon, 30 Jun 1997 16:37:15 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199706301437.QAA21739@dxmint.cern.ch>
Rendered body type: text/plain

>>What I take the 30-tone, 50 baud QPSK modem and scale it down by say
>>a factor of 8 so I get baudrate of 6-7/s, total bandwidth around 300 Hz
>>and total bit rate like 300 bps.
>>Would HF support this low baud rate (this would be phase modulation).
>>I remember somebody saying that on HF the phases are consistant
>>up to about 1/10th of a second ?
>
>So, you want to sell your idea to a bunch of amateurs, where would you like
>to see it fit in above scheme?

I would classify it as a keyboard-to-keyboard/broadcast mode
for weak HF conditions.

I want to put 15 carriers spaced by 15-20 Hz, BPSK modulated (not QPSK,
just BPSK) at 10-15 baud. The bandwidth (200-300 Hz) would fit into an average
CW filter. Then I would do the Walsh function FEC (like in the NEWQPSK)
and interleave. Each symbol (15 bits) would make a BAUDOT character (5 bits)
thus we have 10-15 characters/second (RTTY like).

I feel this is something I can do right away (only a matter of programming
time) and I suspect I would be able to make the Rx lock onto the ongoing
transmition - but still the leading sync. tones would be there to speed
up the signal acquisition at startup.

This thing should be better than the SLOWBPSK because of FEC, interleave
and multi-tone.

I could try as well to double the number of tone and to half the baud rate.
I don't know if this wouldn't be too slow for HF ?
I have as well little idea about the phase noise of average HF tranceivers,
if somebody out there can tell me any numbers please stand up ! :-)

Pawel

78. Re: [HFSIG:2196] dsp56002 evm

From: Bill Carver <bcarver@magiclink.com>
Date: Mon, 30 Jun 1997 10:02:52 -0500 (CDT)
To: hfsig@tapr.org, hfsig@tapr.org
Message-ID: <199706301502.KAA07434@tapr.org>
Rendered body type: text/plain

At 08:38 AM 6/30/97 -0500, Stephen PEARCE wrote:
>Hi
>   Now that the group purchase of these evm s is closed what is
>the best way of obtaining one?
>
>   Thanks
>   Stephen pearce zl1any

Me too. Bill Carver - W7AAZ

79. Re: [HFSIG:2197] Re: Modem patents & reports

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 30 Jun 1997 08:36:12 -0800
To: hfsig@tapr.org
Message-ID: <199706301534.IAA08816@peak.org>
Rendered body type: text/plain

Hi,

>
>On Sun, 29 Jun 1997, Antony Martin wrote:
>
>> > Bergzen Hakan, KR wrote:
>> This is a bit of a bugger as you probably have to pay money to harris
>> if you want to make STANAG 4285 or MIL STD 188 110A modems or do
>> software for it. It's not as though the FEC in these standards is ideal,
>> they've just become accepted standards.
>
>   I don't know about STANAG 4285 but MIL STD 188 110A and even MIL STD
>   118 110C are DoD Standards and fully published.

One can always get your patent pending prior to publication - this way it is
public, yet you have done what is necessry to get the patented secured in
time. That is how academics are instructed to do publish their research
these days. If someone infringes, and your academic institution senses that
there is big bucks to be made, they will make a legal case and pursue the
matter. Does not happen every day, but the are cases.

> Several manufacturers make MIL STD 188 110C modems.

If you are in the manufacturing business, you probably have your own legal sharks deal with patents.

My two cents,

--Johan

BTW - has anyone yet gotten their surplus Harris 39-tone modems operational? I would not mind a doing a test to check them out.

80. RE: [HFSIG:2197] Re: Modem patents & reports

From: "Breon, Roy K (RFC MS-Mail)" <rkb1@rfpo1.rfc.comm.harris.com>
Date: Mon, 30 Jun 1997 11:51:00 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=Harris%l=HARRIS/ROCHESTER/0007E477@rfcmx1.rfc.comm.harris.com>
Rendered body type: text/plain

In response to Walt ---

The MIL STD basically describes the transmitted on-the-air waveform which was developed by Harris but released for the spec. Therefore, anyone is free to develop a modulator without concern for licensing. Demodulation implementation techniques are left to the student and are not described in the spec. Our patented techniques are therefore not described in the MIL STD. Our demodulation technique has several unique features such as excision of up to four interfering signals. This permits us to receive data with negative signal to noise ratios. Some customers intentionally "hide" their signal under a broadcast station or other strong signal!! Other manufacturers cannot use these specific techniques without licensing fees and have implemented the demodulator in another manner. Therefore, there are significant differences in the performance of various manufacturer's "serial" modems. The negative signal to noise ratio performance is not available in our parallel tone modems.

Roy Breon
Harris Corp.

From: Walt DuBose - K5YFW
To: hfsig@tapr.org
Subject: [HFSIG:2197] Re: Modem patents & reports
Date: Monday, June 30, 1997 10:10AM

On Sun, 29 Jun 1997, Antony Martin wrote:

> > Bergzen Hakan, KR wrote:
> This is a bit of a bugger as you probably have to pay money to harris
> if you want to make STANAG 4285 or MIL STD 188 110A modems or do
> software for it. It's not as though the FEC in these standards is ideal,
> they've just become accepted standards.

I don't know about STANAG 4285 but MIL STD 188 110A and even MIL STD 118 110C are DoD Standards and fully published. Several manufacturers make MIL STD 188 110C modems.

I think the 110C Change Notice 2 is the Harris Serial Tone Modem and this probably is restricted.

With all due respect to the Harris Manager who monitors this SIG, I've used both in an almost side-by-side operation and there is to my thoughts only a slight advantage seen using the serial tone modem vs a 39 parallel tone MIL STD modem. For ham radio use, I cannot see the advantage to using the serial tone modem if it is all that more complicated to build/program.

Someone on the SIG has hard copies of MIL-STD-188-110C.

Walt/K5YFW

81. RE: [HFSIG:2188] Re: Serial versus parallel, mainly parallel

From: jalocha@axdele.cern.ch
Date: Mon, 30 Jun 1997 17:49:07 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199706301549.RAA17349@dxmint.cern.ch>
Rendered body type: text/plain

Alexander:
>Constant envelope on each carrier of an OFDM transmission is useless as
>the overall signal is of course not constant envelope. So this can't be
>the reason for using SQPSK.

I think that in a multitone transmition the non-constant envelopes
of the tones do matter because the overall signal will get this envelope
(you hear brrrrr. at the symbol rate) which is not good for the peak/RMS
ratio.

>Furthemore SQPSK (or OQPSK) can't be easily
>demodulated non-coherently.

I demodulate non-coherently anyway - I only look at the phase difference,
never at the absolute phase.

I thought that maybe the SQPSK is to get the soft decisions easier ?

How do you get the soft decisions anyway in differential QPSK ?
Should the weight be proportional to the correlation I get by
multiplying two succesive vectors ?
But I can get very strong (=good ?) correlations for tones jammed
by CW carriers ?

Pawel

82. Re: [HFSIG:2196] dsp56002 evm

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Mon, 30 Jun 1997 11:37:16 -0600 (MDT)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.3.96.970630113530.9764A-100000@murdrum.NMSU.Edu>
Rendered body type: text/plain

On Mon, 30 Jun 1997, Stephen PEARCE wrote:

>   Now that the group purchase of these evm s is closed what is
> the best way of obtaining one?

Purchase one through a distributor that handles Motorola products. You can
find a list of distributors in your are by going to the Motorola WWW site
at http://www.mot.com/SPS/DSP. At the bottom of the page will be a quick
menu that you can pull down and select 'distributors'.

73,
Tim AA5DF

83. Re: QPSK vs. OQPSK (SQPSK)

From: mcdermot@rockdal.aud.alcatel.com (Tom Mcdermott)
Date: Mon, 30 Jun 97 12:38:26 CDT
To: hfsig@tapr.org
Message-ID: <9706301738.AA04692@eagle.aud.alcatel.com>
Rendered body type: text/plain

The general reson for stagering the bit timing
(O=offset, S=staggered) is to prevent the I- and the
Q-channels from chaning at the same time. This eliminates
the instanteanous 180-degreee phase reversal from occuring,
in essense causeing it to occur at two different times.

A narrow-bandwidth filter 'rings-down' to zero amplitude
then builds back up to full amplitude (but opposite
phase) with QPSK (no offset). This imparts amplitude
variations in the envelope. If that signal is hard-limited,
then out-of-band components are generated (those which were
just filtered out). So, QPSK is not too good to put through
a class-C transmit PA if you want to preserve the good
spectral properties possible with QPSK.

OQPSK (or SQPSK) rings-down much less at each phase-change time,
so much less amplitudae variation is imparted on the envelope.
If that signal is hard limited, much less out-of-band energy
is re-created.

- Tom

Tom McDermott
mcdermot@aud.alcatel.com

84. softwave receiver

From: Stephen PEARCE <spearce@ccu1.auckland.ac.nz>
Date: Tue, 1 Jul 1997 07:22:03 +1200 (NZT)
To: hfsig@tapr.org
Message-ID: <199706301922.HAA12512@ccu1.auckland.ac.nz>
Rendered body type: text/plain

Hi
        We have a softwave hf receiver from comfocus.
It has windows based software to run it.
        I would like to write some dos based stuff so that I can use it
with blaster teletype for dcoding hf rtty.
        Does anyone have a programming spec for this board which
would allow me to do this please

        Thanks
        Stephen Pearce zl1any

85. Re: [HFSIG:2191] Re: hsp50016 ddc

From: Antony Martin <anthony.martin@gecm.com>
Date: Mon, 30 Jun 1997 20:35:04 +0100
To: hfsig@tapr.org
Message-ID: <970630203504.3399@oas01.gmcl.co.uk>
Rendered body type: text/plain

> >Hi,..
> >   I am trying to obtain this device to experiment with the construction of an
> >HF receiver along the lines of the qex articles which appeared over the last 3
> >-4 years. Does anyone know where I could purchase it in 1 off quantities or
> >samples .. I have had no luck with the agents here in NZ and Harris have
> not replied.
> >
> >Also looking for one offs of 10 or 12 bit adcs suitable to go with it
> >(such as ad9070 100 MSPS 10 bit adc or Burr Brown ads800 12 bit 40 MSPS)
> >
> >   Any assistance or advice gratefully accepted
> >   Thanks
> >   Stephen Pearce zl1any
> >
You're trying to build a receiver with digital downconversion. A very
interesting project well worth persuing. I wish you happy experimenting.

At which IF do you intend to digitize?

In modern HF practice, the 1st (variable) LO converts to a high 1st IF,
around 45 to 70MHz. This is the Xtal filtered with a roofing filter
(say 12kHz bw). This is followed by conversion to a 2nd IF around 1.4MHz
at which the SSB filters work well. This is followed by a 3rd & final
mix to baseband. Some equipments have 3 IFs.

If you intend digitizing at the 1st or 2nd IF, it is necessary only to
sample at around 72kHz. The wanted BW is +-3kHz, and the unwanted BW
is dictated by the skirts of your roofing filter. This will be around
+-30kHz to 70 or 80dB down (depending on munber of stages). Sampling
at 72kHz, you get a lot of aliasing of the unwanted signals but none
of it falls in the wanted +-3kHz. In fact, +-6kHz is clear of aliases
to allow for DSP filter skirts. Small frequency offsets can also be
implemented in DSP making the 1st LO synth simpler.

The converter may only need to convert in 13.9us but the
sampling time window must be much smaller depending on your IF.

You don't need to resort to application-specific signal processors
here, any 20mips DSP could decimate to baseband. It would help if it
had a buffered serial port or DMA.

A cheap sampling ADC such as the Sipex SP8531 can convert at upto
235kHz but the sampling aperture is 35ns giving a quoted 3dB bw of
13MHz. You should be able to convert a 10.7MHz IF without extra
circuits for a few $$$ or ECUs.

OTOH...
If you're trying to digitise RF directly then 10 or 12 bits just ain't
enough. Not enough dynamic range for out-of-band signals, and not
enough linearity to get a creditable third intercept.

direct RF-to-bits conversion on the HF band is still cloud cuckoo I'm
afraid - A symptom of too many magazine articles syndrome.

Our Marconi Digital receivers & transmitter drives currently use
bandpass sigma-delta conversion at the 2nd IF. The receiver ADC has
a dynamic range of 140dB so the RF & IF has no AGC.
I wouldn't recommend this approach to amateurs as 3rd order
bandpass sigma-delta requires a PhD to understand, I don't claim to
understand it. It also needs downsampling ASICs. This receiver has no
crystal filters or ferrite inductors in the signal path as it would
degrade linearity (!).

? is this interesting for discussion or is it off-topic *?*

- --
A.N. Martin   anthony.martin@gecm.com
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

86. Re: [HFSIG:2201] Re: Modem patents & reports

From: Antony Martin <anthony.martin@gecm.com>
Date: Mon, 30 Jun 1997 22:02:38 +0100
To: hfsig@tapr.org
Message-ID: <970630220238.3399@oas01.gmcl.co.uk>
Rendered body type: text/plain

> In response to Walt ---
>
> The MIL STD basically describes the transmitted on-the-air waveform
> which was developed by Harris but released for the spec. Therefore,
> anyone is free to develop a modulator without concern for licensing.
> Demodulation implementation techniques are left to the student and are
> not described in the spec. Our patented techniques are therefore not
> described in the MIL STD. Our demodulation technique has several unique
<bw-saving snip>

I'm entirely in agreement with Roy Brown's point of view;

In order to meet the specs, any modem not made by harris must transmit
the extra bits of data that help harris modems demodulate. In order
not to use the harris patent, you can't use those bits to demodulate
yourself.

Even if you're a student, if you didn't use the extra bits to help
track or demodulate, you'd be blind or stupid & probably both.

Unless you have developed some method of subverting the laws of
physics, you will not obtain the same performance as a patent
modem if you don't use all the signal power which arrives at
the receiver. You may be able to get so close as makes no odds,
time will tell if this is the case.

There may also be manufacturers who have designed early equipment
that uses extra bits to assist demodulation and who might ignore
bits of the patent.

You might also invent a wonderful encryption system that ensures
these bits are impossibly scrambled and promote it as a world
standard, thus screwing the patentee into the same place as
everyone else.

We're all for standardisation, but all would prefer standards
that are free from manufacturer specific restrictions. The
products of government-funded research should belong to those
that paid for it and on whose behalf it was done; ie. every member
of the population.

(Who funded it in the harris case is not known to me.)

There's no such thing as absolute SNR figures; SNR is always
a comparative term. The magic spell '0dB SNR' has meaning
only to he who casts it. It all depends what you take as
your signal BW and your noise BW and how much knowledge of the
modulation you take into account in your analysis of SNR.
All receivers can be analysed to yeild a true effective noise
bandwidth, even if the modulation is spread over a wider BW.

As an example, my modems hopping across 1MHz BW work fine in
- -20dB SNR! That's rms signal power in 1MHz over rms noise power
in 1MHz. See my point? Just don't ask how many interferers it
can cope with in 1MHz. I don't know & I dread to think.

(above discussion now terminated to avoid it becoming a
 spam)

- --
A.N. Martin   anthony.martin@gecm.com
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

87. unsubsribe hfsig

From: RAYMOND CZYZEWSKI <wa2sei@nac.net>
Date: Mon, 30 Jun 1997 19:09:47 -0400
To: hfsig@tapr.org
Message-ID: <33B83CBB.296@nac.net>
Rendered body type: text/plain

[Empty message body]

88. unsubsribe hfsig

From: "James Brown" <jbrown2@san.rr.com>
Date: Mon, 30 Jun 1997 19:07:00 -0700
To: "hgsig" <hfsig@tapr.org>
Message-ID: <199707010209.TAA29512@mail.san.rr.com>
Rendered body type: text/plain

[Empty message body]

89. Re: [HFSIG:2199] Re: dsp56002 evm

From: "Steven R. Bible" <srbible@gate.net>
Date: Mon, 30 Jun 1997 22:10:55 -0400
To: hfsig@tapr.org
Message-ID: <3.0.1.32.19970630221055.007a24b0@pop.gate.net>
Rendered body type: text/plain

At 10:06 AM 6/30/97 -0500, you wrote:
>At 08:38 AM 6/30/97 -0500, Stephen PEARCE wrote:
>>Hi
>>   Now that the group purchase of these evm s is closed what is
>>the best way of obtaining one?
>>
>>   Thanks
>>   Stephen pearce zl1any
>
>Me too. Bill Carver - W7AAZ

Before the TAPR purchase I bought my DSP56002EVM from Newark. Their URL is

    http://www.newark.com/

In fact a search of their web catalog produced the following

Newark Pt No   Mfgr Pt No   Manufacturer Available Buy Mult U of M
Min Buy DSP56002EVM   DSP56002EVM MOTOROLA   4   1
EA0001 1

Cat 115 Page No Cat 114 Page No Status Catalog Status
889   20   new   available, in catalog

Two years ago I paid US$149.00.

- Steve, N7HPR
(n7hpr@tapr.org)

July 1997

Original source file: hfsig.9707
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9707

1. Offset QPSK

From: Alan Bloom <alanb@manta.sr.hp.com>
Date: Mon, 30 Jun 1997 22:04:11 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199707010504.AA150343452@manta.sr.hp.com>
Rendered body type: text/plain

Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de> wrote:

>Johan Forrer wrote:
>>
>> Pawel,
>>
>> >>Our experimental efforts are using m-DQPSK. Phil Karn (the famous KA9Q
>> >> - Mr. TCP/IP) is suggesting the offset form (SQPSK).
>> >
>> >Could someone please remind me shortly why the shifted form is better ?
>> >I am building up a new parallel tone modem and I want to use
>> >the SQPSK but I must say I don't know why should I use it ? :-)
>>
>> It yields a constant signal envelope as far as I can remember, but that has
>> some quirks that confounds carrier/clock recovery. Perhaps Phil can provide
>> a better description.

Offset QPSK is not truely constant-envelope. But because it avoids
trajectories that cross near the origin of the I/Q plane (i.e. the zero-
power point), the peak-to-average power ratio is lower than for regular
QPSK, which eases linearity requirements for the transmitter power
amplifier. PI/4 DQPSK is used for the same reason.

>Constant envelope on each carrier of an OFDM transmission is useless as
>the overall signal is of course not constant envelope. So this can't be
>the reason for using SQPSK.

Right. That's why IS-95 CDMA uses OQPSK for the mobile stations but
regular QPSK for the base stations, which transmit multiple channels.
(Of course we are talking about multiple carriers rather than multiple
code channels, but the principle is the same.)

jalocha@axdele.cern.ch (Pawel) wrote:

>I think that in a multitone transmition the non-constant envelopes
>of the tones do matter because the overall signal will get this envelope
>(you hear brrrrr. at the symbol rate) which is not good for the peak/RMS
>ratio.

But the maximum/minimum power ratio due to the multiple tones (equal
to 10*log(# of tones) dB) is far greater than the symbol-rate amplitude
variation of Offset QPSK (a couple dB or so, depending on the
pre-modulation filter used).

Alan Bloom N1AL

2. Keyboard-to-Keyboard Bandwidth

From: Alan Bloom <alanb@manta.sr.hp.com>
Date: Mon, 30 Jun 1997 22:08:03 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199707010508.AA150413683@manta.sr.hp.com>
Rendered body type: text/plain

forrerj@peak.org (Johan Forrer) wrote:

>In my opinion, there is a need for several digital modes that I think will
>excite people - just a few ideas here:
>
>A) Keyboard to keyboard conversations (I) (for good signals:) As good or
>better than Pactor II for speed/robustness/interactiveness. Bandwidth approx
>500 Hz. Maximum throughput roughly 900 bps.

Why is 900 bps needed for keyboard-to-keyboard conversations? That's
equivalent to about 900 wpm, which is a little faster than most of us
can type!

Alan Bloom N1AL

3. IEE conference

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Tue, 01 Jul 1997 09:56:33 +0200
To: hfsig@tapr.org
Message-ID: <33B8B831.40839B9@zeus.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

Hi!

I'm just curious. Who is going to the IEE conference on HF Radio Systems
and Techniques (next week in Nottingham)?

73' Alexander

--
-------------------------------+---------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| technik u. Elektroakustik | Fax : +49-6151-165545
|
| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de
|
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
-------------------------------+---------------------------------------

4. The NordicHF homepage

From: "Bergzen Hakan, KR" <hakan.bergzen@enator.se>
Date: Tue, 1 Jul 1997 10:56:00 +0100
To: hfsig <hfsig@tapr.org>
Message-ID: <33BE541F@noak>
Rendered body type: text/plain

Hello all,

Every third year an international HF conference is held on a small island in
the middle of the Baltic Sea. It is called the Nordic HF conference, and the
next event will be HF98. Even though it is mainly directed towards
professional use of 'Earths natural satellite' I thought it might be
appropriate to inform you that the NordicHF web site has once again
re-awakened. The web site address is:
http://www.communications.enator.se/NordicHF/index.htm
(default.htm works just as well)

73 de Hakan

5. Re: [HFSIG:2213] Keyboard-to-Keyboard Bandwidth

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Tue, 1 Jul 1997 07:49:57 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.970701073244.20663A-100000@www.kelly-afb.org>
Rendered body type: text/plain

On Tue, 1 Jul 1997, Alan Bloom wrote:

> >A) Keyboard to keyboard conversations (I) (for good signals:) As good or
> >better than Pactor II for speed/robustness/interactiveness. Bandwidth approx
> >500 Hz. Maximum throughput roughly 900 bps.
>
> Why is 900 bps needed for keyboard-to-keyboard conversations? That's
> equivalent to about 900 wpm, which is a little faster than most of us
> can type!
>
> Alan Bloom N1AL

I thought that 900 bps was about 120 bytes per second (or
characters) and a word measured as 5 characters? That would be
about 24 WPS or 1440 WPM.

But most of the time while working keyboard CW I have many canned
messages and also send short files so 24 WPM is slow. Why not
2400 bps or 3600 WPM? Or 1200 bps = 1800 WPM?

SMTP on VHF works fairly well at 1200 baud so I would assume that 1200 bps would work for SMTP on HF and then we woldn't have to change speeds for two important modes.

I must say I'd rather see point-to-point SMTP rather than keyboard-to-keyboard.

I've probably got my math wrong...if so, would someone set me straight please.

Thanks,

Walt/K5YFW

6. Re: [HFSIG:2213] Keyboard-to-Keyboard Bandwidth

From: Lyle Johnson <lylej@azstarnet.com>
Date: Tue, 01 Jul 1997 06:53:57 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19970701065204.006ef094@pop.azstarnet.com>
Rendered body type: text/plain

At 12:13 AM 7/1/97 -0500, you wrote:
>forrerj@peak.org (Johan Forrer) wrote:
>
>>In my opinion, there is a need for several digital modes that I think will
>>excite people - just a few ideas here:
>>
>>A) Keyboard to keyboard conversations (I) (for good signals:) As good or
>>better than Pactor II for speed/robustness/interactiveness. Bandwidth approx
>>500 Hz. Maximum throughput roughly 900 bps.
>
>Why is 900 bps needed for keyboard-to-keyboard conversations? That's
>equivalent to about 900 wpm, which is a little faster than most of us
>can type!

If the speed of synchronization is quick enough, and there is stattion ID informaiton in every transmission, this sort of speed can result in a bursty operation allowing multiple (perhaps 2 or 3) QSOs to share the same channel. Send the entire line when the typist hits <enter>, and send it quickly...

No, I'm not talking AX.25, just thinking out loud. Using GPS to synchronize the start of a second, TDM "slots" could be arranged to help allow for this sort of thing.

Cheers,

Lyle

7. Re: [HFSIG:2213] Keyboard-to-Keyboard Bandwidth

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 01 Jul 1997 07:54:02 -0800
To: hfsig@tapr.org
Message-ID: <199707011452.HAA25018@peak.org>
Rendered body type: text/plain

Hi Alan,

>forrerj@peak.org (Johan Forrer) wrote:
>
>>In my opinion, there is a need for several digital modes that I think will
>>excite people - just a few ideas here:
>>
>>A) Keyboard to keyboard conversations (I) (for good signals:) As good or
>>better than Pactor II for speed/robustness/interactiveness. Bandwidth approx
>>500 Hz. Maximum throughput roughly 900 bps.
>
>Why is 900 bps needed for keyboard-to-keyboard conversations? That's
>equivalent to about 900 wpm, which is a little faster than most of us
>can type!
>
>Alan Bloom N1AL
>
>

Right - when Pactor I started overtaking AMTOR popularity, folks started going beyond typing text - sending pretty pictures, long brag tapes with extended character sets to use color, cross-band HF/VHF work and all sorts of canned applications. I attribute this as one reason why one hear very

little AMTOR nowadays on the amateur bands. If we look at the next standard,
900 bps sounds reasonable, something like Pactor II, then perhaps we will
perhaps see folks playing with real time person to person digitized speech,
or perhaps digital image transmission. The concept of keyboard-to-keyboard
perhaps is merely metaphorical - I trust that folks understood what was
meant - the other side of the coin for CSMA.

What comes rather close to real keyboard to keyboard work is narrow
bandwidth BPSK. As I was printing slow BPSK the other evening on 40m, which
reminded me of the days when I used to sit and watch my model 28 pound out
the characters one by one. The BPSK was printing even when the signal faded
away completely or there was SSB splatter obscuring the signal. This is
typical for the other class keyboard to keyboard work; that of weak signal
communications.

--Johan

8. Re: [HFSIG:2218] Re: Keyboard-to-Keyboard Bandwidth

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Tue, 1 Jul 1997 10:31:47 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.970701102936.20663C-100000@www.kelly-afb.org>
Rendered body type: text/plain

Johan,

I have been told by "Industry Experts" that you needed 2400 bps to do
realtime digitized speech and 4800 bps if you wanted it to sound kinda
normal. Of course that was using LCP10 and 16/39 parallel tone modems.

Walt/K5YFW

On Tue, 1 Jul 1997, Johan Forrer wrote:

> Hi Alan,
>
> >forrerj@peak.org (Johan Forrer) wrote:
> >
> >>In my opinion, there is a need for several digital modes that I think will
> >>excite people - just a few ideas here:
> >>
> >>A) Keyboard to keyboard conversations (I) (for good signals:) As good or
> >>better than Pactor II for speed/robustness/interactiveness. Bandwidth approx
> >>500 Hz. Maximum throughput roughly 900 bps.
> >
> >Why is 900 bps needed for keyboard-to-keyboard conversations? That's
> >equivalent to about 900 wpm, which is a little faster than most of us
> >can type!
> >
> >Alan Bloom N1AL
> >
> >
>
> Right - when Pactor I started overtaking AMTOR popularity, folks started
> going beyond typing text - sending pretty pictures, long brag tapes with
> extended character sets to use color, cross-band HF/VHF work and all sorts
> of canned applications. I attribute this as one reason why one hear very
> little AMTOR nowadays on the amateur bands. If we look at the next standard,
> 900 bps sounds reasonable, something like Pactor II, then perhaps we will
> perhaps see folks playing with real time person to person digitized speech,
> or perhaps digital image transmission. The concept of keyboard-to-keyboard
> perhaps is merely metaphorical - I trust that folks understood what was
> meant - the other side of the coin for CSMA.
>
> What comes rather close to real keyboard to keyboard work is narrow
> bandwidth BPSK. As I was printing slow BPSK the other evening on 40m, which
> reminded me of the days when I used to sit and watch my model 28 pound out
> the characters one by one. The BPSK was printing even when the signal faded
> away completely or there was SSB splatter obscuring the signal. This is
> typical for the other class keyboard to keyboard work; that of weak signal
> communications.
>
> --Johan
>
>

9. BPSK & Parallel modem tests status:

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 01 Jul 1997 08:42:48 -0800
To: HFSIG@tapr.org
Message-ID: <199707011541.IAA02080@peak.org>
Rendered body type: text/plain

Hi all,

Most of you are aware that there is some modem testing activity going on:

BPSK
----

1) With a group of amateurs monitoring in the UK and the Netherlands using sensitive narrow-band signal detection software, the KC7WW beacon that operated on 14070 (carrier) was not detected. I guess if there is no propagation, there is no signal to work with, regardless of what you do. The experiment will now reverse direction with PA0IHD transmitting a beacon for a week on 14070, so this is an opportunity to see if we succeed this time.

Several reports were received from US amateurs that heard the KC7WW beacon. This is good news - if you can hear the signal, your copy will be excellent. Also, several are in the process to get their EVM interfaces sorted out. Just a reminder; to do slowspeed BPSK with the EVM, you do not need an elaborate interface - just pipe audio to and from the EVM and put your radio in VOX mode. Be careful to turn your power output way down because this is 100% duty cycle and you do not want to burn out your finals.

2) I have had good success with the VE2IQ Coherent software to print BPSK traffic. Signals from the west coast is very strong here and copy is 100%. I also managed to copy about 5% from VE2IQ - a signal that was there once you knew what to listen for. If this was CW, it would have been nil copy for me - I am impressed.
To participate in this project, you need to a build simple sigma-delta A/D convertor that plugs into your serial port. The hardware was featured in QST some time ago, however, the schematic for the A/D and BPSK and software is available for downloading from VE2IQ's web page:
http://w3.ietc.ca/home/bill/bbs.htm

The VE2IQ software allows one to set your symbol length to any 1.5ms multiple - basically you need to acquire an integer number of 800 Hz audio cycles. So any symbol length from say 5ms to 1000ms would be possible. The longer the symbol time, of course the narrower and more critical tuning-in procedures become. We played with 30ms and 25ms sysmbols on 40m that seemed to work very well. This software is quite useful for several kinds of weak signal work; EME and HF are good possibilities. Unfortunately, none of these setting is compatible with the EVM-based BPSK work - that needs 32ms symbols, which is not a multiple of 1.25ms. I do have good intentions to modify the EVM BPSK code to do that, but only when and if I get the time (no promises).

Parallel-tone 3000+ BPS NEWQPSK
----

3) I have only had two responses to a call for participation. One from the US and one from Finland. This modem needs good signals (ca. S6), so our chances to make it to Finland is quite slim, but we decided to monitor the CW beacons and see when there is an opening.

The situation with the US with testing this modem is a bit dissapointing; I have heard about the need for such a high speed CSMA modem - here it is folks! Were are you?

We still are in need for participants in testing these modems, please e-mail me if you are interested. These are really fun toys to play with.

--Johan, KC7WW
  July 1, 1997

10. Re: [HFSIG:2207] US Patent 4,365,338

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 01 Jul 1997 10:03:57 -0800
To: hfsig@tapr.org
Message-ID: <199707011702.KAA14244@peak.org>
Rendered body type: text/plain

Hello Kerry,

First, thank you very much for preparing the document for us. I printed it
on my deskjet printer and it is quite legible, though no fun to read :-).

Most of us find it of great value, however, you may be getting some flack
because of the size of the posting on this list. The other thing is that you
will pestered repeatedly for a copy because the document was not taken up in
the archives - it was a one-time posting, only to current list subscribers.

One suggestion of how to handle this would be to let TAPR keep the copy on
the server for downloading. A short posting on the list of its whereabouts
would then be sufficient.
Just let me know how I can assist with something like this.

Thanks much,

--Johan
(HFSIG moderator)

>Attached is US Patent 4,365,338, downloaded as
>26 separate bit-mapped images; then imported
>into a Word document to facilitate printing as
>a single document; then zipped to fit into a
>smaller space.Content-Type: application/octet-stream; name="Pat4365338.zip"

>Content-Description: Pat4365338.zip (WinZip File)
>Content-Disposition: attachment; filename="Pat4365338.zip"
>
>Attachment Converted: C:\HIGHWAY\MAIL3\Pat43653.zip
>

11. Synchronisation for OFDM ..

From: "Frank Knoll" <fknoll@ibm.net>
Date: Tue, 01 Jul 97 20:53:35 +0100
To: "HFSIG" <hfsig@tapr.org>
Message-ID: <199707011856.SAA110314@pop01.ny.us.ibm.net>
Rendered body type: text/plain

Hello Folks,

has anybody experience about synchronisation (Time and Frequency) with OFDM ???
best regards
Frank Knoll

==
* Frank Knoll (DL2FCH)
* Spessartstr. 12
* D-63571 Gelnhausen-Meerholz
* Germany
* Tel.: +49 6051 969116
* E-Mail: fknoll@ibm.net   or dl2fch@amsat.org
==

12. BPSK tests & becaons

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Tue, 01 Jul 1997 21:29:00 -0400
To: hfsig@tapr.org
Message-ID: <199707020128.VAA26302@aus-a.mp.campus.mci.net>
Rendered body type: text/plain

Johan,

I assume PA0IHD's beacon will be on the same sked (2000 - 0200 utc).

Given that propagation conditions are not all that great, would it be worth
trying a beacon on 40 mtrs? All I've heard on 14070.0 so far has been
either very strong pactor station or a dead band.

Don, W4GWC

13. Re: [HFSIG:2221] Re: US Patent 4,365,338

From: "Kerry Berland" <kberland@ix.netcom.com>
Date: Tue, 1 Jul 1997 20:14:46 -0500
To: <hfsig@tapr.org>

Message-ID: <199707020138.UAA17282@dfw-ix10.ix.netcom.com>
Rendered body type: text/plain

Sorry to anyone whose email was clogged by
the length of this. Perhaps this was poor
judgment. Thanks Johann for your
suggestions. Will consult before doing
this again. Anyone who needs a copy,
you could email me directly.

> From: Johan Forrer <forrerj@peak.org>
> To: hfsig@tapr.org
> Subject: [HFSIG:2221] Re: US Patent 4,365,338
> Date: Tuesday, July 01, 1997 12:12 PM
>
> Hello Kerry,
>
> First, thank you very much for preparing the document for us. I printed
it
> on my deskjet printer and it is quite legible, though no fun to read :-).

>
> Most of us find it of great value, however, you may be getting some flack
> because of the size of the posting on this list. The other thing is that
you
> will pestered repeatedly for a copy because the document was not taken up
in
> the archives - it was a one-time posting, only to current list
subscribers.
>

14. Re: [HFSIG:2211] Re: dsp56002 evm

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Tue, 1 Jul 1997 19:50:00 -0600 (MDT)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.3.96.970701194810.6328A-100000@murdrum.NMSU.Edu>
Rendered body type: text/plain

I understand that the $149 for the EVM is a 'suggested retail'. If a
distributor is asking more, check around.

Tim

On Mon, 30 Jun 1997, Steven R. Bible wrote:

> At 10:06 AM 6/30/97 -0500, you wrote:
> >At 08:38 AM 6/30/97 -0500, Stephen PEARCE wrote:
> >>Hi
> >>   Now that the group purchase of these evm s is closed what is
> >>the best way of obtaining one?
> >>
> >>   Thanks
> >>   Stephen pearce zl1any
> >
> >Me too. Bill Carver - W7AAZ
>
>
> Before the TAPR purchase I bought my DSP56002EVM from Newark. Their URL is
>
>   http://www.newark.com/
>
> In fact a search of their web catalog produced the following
>
> Newark Pt No   Mfgr Pt No   Manufacturer Available Buy Mult U of M
> Min Buy DSP56002EVM   DSP56002EVM MOTOROLA   4   1
> EA0001 1
>
> Cat 115 Page No Cat 114 Page No Status Catalog Status
> 889   20   new   available, in catalog
>
> Two years ago I paid US$149.00.
>
>
>
> - Steve, N7HPR
> (n7hpr@tapr.org)
>

15. unsubscribe hfsig

From: Ed Lyons <eel@ipass.net>
Date: Tue, 01 Jul 1997 22:12:54 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970702021254.0069feb4@ipass.net>
Rendered body type: text/plain

[Empty message body]

16. unsubscribe hfsig

From: Bill Carver <bcarver@magiclink.com>
Date: Tue, 1 Jul 1997 23:17:03 -0500 (CDT)
To: <hfsig@tapr.org>
Message-ID: <199707020417.XAA06950@tapr.org>
Rendered body type: text/plain

[Empty message body]

17. Re: [HFSIG:2206] Re: hsp50016 ddc

From: Larry Gadallah <larry@ve6vq.ampr.org>
Date: Tue, 1 Jul 97 21:40:59 -0700
To: hfsig@tapr.org
Message-ID: <199707020441.VAA04588@ve6vq.ampr.org>
Rendered body type: text/plain

Stephen Pearce wrote:
> > >Hi,.. I am trying to obtain this device to experiment with the
> > >construction of an HF receiver along the lines of the qex
> > >articles which appeared over the last 3 -4 years. Does anyone
> > >know where I could purchase it in 1 off quantities or samples ..
> > >I have had no luck with the agents here in NZ and Harris have
> > not replied.
> > >
> > >Also looking for one offs of 10 or 12 bit adcs suitable to go
> > >with it (such as ad9070 100 MSPS 10 bit adc or Burr Brown ads800
> > >12 bit 40 MSPS)
> > >

I am quite interested in the referenced QEX articles. Does anyone
have a list of which issues of QEX they appeared in?

Thanks in advance,

Larry Gadallah, VE6VQ/K6   larry@ve6vq.ampr.org
Walnut Creek, California   +1 510 943-2164
Key fingerprint = D6 79 5D 9D 41 27 74 03 68 FD D7 F3 86 68 EB A5

18. [HFSIG:2213] Keyboard-to-Keyboard Bandwidth

From: kl7ev@actis.polarnet.com
Date: 1 Jul 1997 18:24:36 -0800 (AST)
To: hfsig@tapr.org
Message-ID: <199707020534.AAA19131@tapr.org>
Rendered body type: text/plain

Alan Bloom wrote:

HF>Why is 900 bps needed for keyboard-to-keyboard conversations? That's
HF>equivalent to about 900 wpm, which is a little faster than most of us
HF>can type!

HF>Alan Bloom N1AL

I think you need to check your math, 900bps equates to around 80 to 100
Characters per second or roughly 16 word per minute. Well within typing
speed!

       - Greg -
----
kl7ev
----

19. [HFSIG:2229] Keyboard-to-Keyboard Bandwidth

From: kl7ev@actis.polarnet.com
Date: 1 Jul 1997 22:22:12 -0800 (AST)
To: hfsig@tapr.org
Message-ID: <199707020720.CAA04922@tapr.org>
Rendered body type: text/plain

HF>Alan Bloom wrote:

HF>HF>Why is 900 bps needed for keyboard-to-keyboard conversations? That's
HF>HF>equivalent to about 900 wpm, which is a little faster than most of us
HF>HF>can type!

HF>HF>Alan Bloom N1AL

Well, I just checked my own math and decided I was wrong instead. Hi!
900bps just doesn't seem like 900wpm. Guess I was wrong,

- Greg -
---
kl7ev
---

20. [HFSIG:2229] Keyboard-to-Keyboard Bandwidth

From: kl7ev@actis.polarnet.com
Date: 1 Jul 1997 22:22:14 -0800 (AST)
To: hfsig@tapr.org
Message-ID: <199707020720.CAA04944@tapr.org>
Rendered body type: text/plain

HF>Alan Bloom wrote:

HF>HF>Why is 900 bps needed for keyboard-to-keyboard conversations? That's
HF>HF>equivalent to about 900 wpm, which is a little faster than most of us
HF>HF>can type!

HF>HF>Alan Bloom N1AL

Well, I just checked my own math and decided I was wrong instead. Hi!
900bps just doesn't seem like 900wpm. Guess I was wrong,

- Greg -
---
kl7ev
---

21. Interesting New Book on HF Radio Comms

From: Rolf Sommerhalder <rs@ife.ee.ethz.ch>
Date: Wed, 2 Jul 1997 14:46:02 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199707021246.OAA02088@makeba.ee.ethz.ch>
Rendered body type: text/plain

Just discovered in our library...
(see http://www.artech-house.com/artprogs/detail_disp?0-89006-815-1)

Advanced High-Frequency Radio Communication
Eric E. Johnson, Robert I. Desourdis, Greg Earle, Stephen Cook, Jens C.
Ostergaard; 1997
ISBN: 0-89006-815-1
376 pp.

This up-to-date, comprehensive reference and planner□s guide presents an in-depth
description of automated HF communications, considers the many phenomena that
affect HF radio, and offers penetrating insight into new and future HF techniques,
including:

- New information on automatic link establishment (ALE) technology
- Advanced variable-transmission rate modems
- Wideband (1 MHz) HF communications services
- Automated HF networks as an extension of the Internet

Helpful chapter-end examples, 100 illustrations, and 135 equations make this an
invaluable guide for communications engineers involved with the manufacture of HF
radio and ancillary equipment. It draws on experts□ knowledge to offer new
approaches for establishing and maintaining an automated HF network, based on
industry and governmental standards. It also serves as an excellent tutorial for
graduate students and systems-level planners.

Contents: Introduction. HF Radio Propagation and Noise. HF Radio Communications.
HF Radio Networks. Future Developments.

73,
Rolf HB9CWP

22. Re: [HFSIG:2221] Re: US Patent 4,365,338

From: dona@tridelta.com (Don Adams)
Date: Wed, 2 Jul 1997 09:18:45 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199707021318.JAA13348@sparcy.tridelta.com>
Rendered body type: text/plain

Hello Kerry,

>First, thank you very much for preparing the document for us. I printed it
>on my deskjet printer and it is quite legible, though no fun to read :-).

I agree

>Most of us find it of great value, however, you may be getting some flack
>because of the size of the posting on this list. The other thing is that you
>will pestered repeatedly for a copy because the document was not taken up in
>the archives - it was a one-time posting, only to current list subscribers.

I disagree. I was glad to get the copy with the e-mail. I find it inconvient to
have log into my server and try to find and download a file. If I don't want
the attachment, it's easy to delete it. Easier that trying to go out on the net
and get it. Also many people only have e-mail (such as juno).

>One suggestion of how to handle this would be to let TAPR keep the copy on
>the server for downloading. A short posting on the list of its whereabouts
>would then be sufficient.

I agree

over and out with my 2 cents.....Don wa8qzz

23. Re: [HFSIG:2207] US Patent 4,365,338

From: nt1g@bitwise.net (Skip Flem)
Date: Wed, 02 Jul 1997 11:17:30
To: hfsig@tapr.org
Message-ID: <02199730.Amiga@mailbox.bitwise.net>
Rendered body type: text/plain

> Work:SETUP/IBrowser2/mailout/1997.07.02.001.AM

>
> Attached is US Patent 4,365,338, downloaded as
> 26 separate bit-mapped images; then imported
> into a Word document to facilitate printing as
> a single document; then zipped to fit into a
> smaller space.
>
>
>
>

--
Skip Flem, NT1G <nt1g@bitwise.net> Hull, Massachusetts 02045 U.S.A.

24. THE MOTHER OF ALL TURDS:K.BERLAND

From: nt1g@bitwise.net (Skip Flem)
Date: Wed, 02 Jul 1997 11:27:45
To: hfsig@tapr.org
Message-ID: <02199745.Amiga@mailbox.bitwise.net>
Rendered body type: text/plain

> Attached is US Patent 4,365,338, downloaded as
> 26 separate bit-mapped images; then imported
> into a Word document to facilitate printing as
> a single document; then zipped to fit into a
> smaller space.
>
Didn't they invent the WWW to PREVENT POSTings like this???

1 meg mail messages just SCREW UP my mailer.

AND I THOUGHT MIKE TYSON JUST CLINCHED "JERK OF THE WEEK".
(I have a bunch more...)

--

25. Re: [HFSIG:2235] Why all the bovver over one patent post?

From: Antony Martin <anthony.martin@gecm.com>
Date: Wed, 02 Jul 1997 19:37:16 +0100
To: hfsig@tapr.org
Message-ID: <970702193716.29615@oas01.gmcl.co.uk>
Rendered body type: text/plain

> > Attached is US Patent 4,365,338, downloaded as
> > 26 separate bit-mapped images; then imported
> > into a Word document to facilitate printing as
> > a single document; then zipped to fit into a
> > smaller space.
> >
> Didn't they invent the WWW to PREVENT POSTings like this???
>
> 1 meg mail messages just SCREW UP my mailer.
>
I thought mailers that could only cope with 64k files went
obsolete years ago. How much RAM have you got in your machine?

Get yourself a mail reader that works. There's no excuse.
I hadn't noticed the size - it's LESS than 1 Meg, 929279 bytes.

Amigas have a 680x0 which has NO 64k limits. Why do Amigas have a problem?

My only complaint is that the version of Word required is
not defined; and word output is inclined to be non-portable,
even between versions of Word. You can't load a document into
an earlier version of Word, even if it doesn't use the features
of the later version. The Gates plan for deliberate obsolescence.

I like PDF, Postscript or GIFs. They can all be viewed with FREE
software, on any platform.

A happy UniX user :-)

ps. Word runs much better on unix; when it crashes you don't
have to reboot the machine!

- ------
Word is a TM of Microsoft; Unix is TM of AT&T; Amiga is TM of Commodore
PDF belongs to Adobe; Postscript too; And Compuserve has the rights to
GIFs. 'WWW' isn't anybody's but they think 'Internet' is. Gates are an
invention used to obstruct movement through gaps in fences. Existence
of persons of this name living or dead is coincidental; And IPR is a pain.
Anyone I've forgotten will probably do a search & find this message and sue
me.

26. EVM56K for sale, again

From: Kory Hamzeh <kory@avatar.com>
Date: Wed, 2 Jul 1997 15:23:32 -0700 (PDT)
To: dsp@tapr.org, hfsig@tapr.org
Message-ID: <Pine.BSI.3.91.970702151522.11415A-100000@avatar.avatar.com>
Rendered body type: text/plain

Sorry for posting this the second time. The buyer who was hot on buying
this from me stopped responding all of the sudden.

I have an EVM56K here which is gathering dust. It includes all of the docs
and software. This was not through the TAPR purchase. I would like to get
$125 + shipping for it. It has been socketed for the flash chip.

Thanks,
Kory
AC6RN

27. [HFSIG:2226] unsubscribe hfsig

From: Ed Lyons <eel@ipass.net>
Date: Wed, 02 Jul 1997 18:34:34 -0400
To: hfsig@tapr.org

Message-ID: <1.5.4.32.19970702223434.0069ff24@ipass.net>
Rendered body type: text/plain

[Empty message body]

28. need help

From: Aureusgrp@aol.com (by way of Greg Jones, WD5IVD)
Date: Wed, 2 Jul 1997 19:08:32 -0500
To: "NETSIG list mailing"<netsig@tapr.org>,
" Spread Spectrum "<ss@tapr.org>,
"HF SIG list mailing"<hfsig@tapr.org>
Message-ID: <v03020912afe09dde0102@[208.134.134.40]>
Rendered body type: text/plain

For those that might be interested.

Greg

Hi. We're specialized recruiters and could really use you help. A client of
ours has a high profile DAMA program and has a critical need for one or more
DAMA engineers. These are exciting positions in design, development, and
systems integration. Knowledge of approriate mil-stds is part of the picture
too. Who do you know that I can talk to? Thank you so much for any help you
can give us. Nancy Ulbert, CPC aureusgrp@aol.com or 800-289-4044.

29. Re: need help

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Thu, 3 Jul 1997 00:13:02 -0500
To: ss@tapr.org
Cc: "HF SIG list mailing"<hfsig@tapr.org>,
"NETSIG list mailing"<netsig@tapr.org>
Message-ID: <v0302090aafe0e51dc2bd@[208.134.134.40]>
Rendered body type: text/plain

Thanks for letting us know Lyle. Hopefully someone will make sure that
TAPR gets the finders fee if they respond. I know we could sure use it.

I sure didn't know.

Cheers - Greg

>At 07:20 PM 7/2/97 -0500, you wrote:
>>For those that might be interested.
>>
>>Greg
>>
>>------
>>
>>Hi. We're specialized recruiters and could really use you help. A client of
>>ours has a high profile DAMA program and has a critical need for one or more
>>DAMA engineers. These are exciting positions in design, development, and
>>systems integration. Knowledge of approriate mil-stds is part of the picture
>>too. Who do you know that I can talk to? Thank you so much for any help you
>>can give us. Nancy Ulbert, CPC aureusgrp@aol.com or 800-289-4044.
>>
>Greg,
>
>These guys get big bucks to recruit, so if anyone repsonds, through this
>message, please inform the recruiter that 25% of his fee goes to TAPR or
>you don't take the job :-)
>
>(The fees are often 40% to 60% of the annual compensation of the person
>recruited - I know, we just went through one to get a person we needed...)
>
>Cheers,
>
>Lyle

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd
-----

30. RE: [HFSIG:2235] THE MOTHER OF ALL TURDS:K.BERLAND

From: "Fortney, James T." <FortneyJ@ix.netcom.com>
Date: Wed, 2 Jul 97 22:51:28 PDT
To: hfsig@tapr.org
Message-ID: <Chameleon.4.01.3B.970702230059.fortneyj@tok-ca6-27.ix.netcom.com>
Rendered body type: text/plain

REFERENCE THE 1.4MB mail item.

>Didn't they invent the WWW to PREVENT POSTings like this???
>
>1 meg mail messages just SCREW UP my mailer.

I LOST over 20 messages (and about 5 hours) trying to deal with the problems
created by this message. I would suggest to TAPR that a size parameter needs
to be implemented as part of the re-mailer to protect us from "insensitive"
acts like the one we experienced here.

>AND I THOUGHT MIKE TYSON JUST CLINCHED "JERK OF THE WEEK".
>(I have a bunch more...)

I have to assume the sender was ignorant of the grief he was creating for
the hundreds of Users who still have not recovered.

- JimF   K6IYK

James T. Fortney
E-mail: Jim@Fortney.org
    or: Jim@Fortney.com
    or: FortneyJ@ix.netcom.com
Voice: 805.491.3916 or 1.500.FORTNEY
AX.25:   K6IYK@K6IYK.#SCA.CA.USA.NOAM
MARSAMTS: AAA9CS@AT9TCS.AMARS.DOD
FAX: 805.491.0319
Snail: P.O. Box 3419
          Camarillo, CA 93011-3419
----

31. Re: [HFSIG:2202] Re: Serial versus parallel, mainly parallel

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Thu, 03 Jul 1997 11:15:30 +0200
To: hfsig@tapr.org
Message-ID: <33BB6DB2.5D3B43B0@zeus.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

jalocha@axdele.cern.ch wrote:
>
> Alexander:
> >Constant envelope on each carrier of an OFDM transmission is useless as
> >the overall signal is of course not constant envelope. So this can't be
> >the reason for using SQPSK.
>
> I think that in a multitone transmition the non-constant envelopes
> of the tones do matter because the overall signal will get this envelope
> (you hear brrrrr. at the symbol rate) which is not good for the peak/RMS
> ratio.

Yes, this is a problem. A collegue here analysied this in his PhD
thesis. In general you can say that if you try to keep an (almost)
constant envelope for an OFDM signal you have to put some redundancy in
the signal. And usually it is better to use this redundancy for coding
and simply do some soft clipping a live with the distortion.

>
> >Furthemore SQPSK (or OQPSK) can't be easily
> >demodulated non-coherently.
>
> I demodulate non-coherently anyway - I only look at the phase difference,
> never at the absolute phase.
>
> I thought that maybe the SQPSK is to get the soft decisions easier ?

Not as far as I know.
>
> How do you get the soft decisions anyway in differential QPSK ?
> Should the weight be proportional to the correlation I get by
> multiplying two succesive vectors ?
> But I can get very strong (=good ?) correlations for tones jammed
> by CW carriers ?

Let "s(n)" be the possibly transmitted signal at time t=nT and "r(n)"
the received one.
Usually your metric is (s(n)-CSI*r(n))^2 with CSI the channel state
information. For differential non-coherent demodulation we take r*(n-1)
(the last received symbol) as our "CSI", so your metric becomes (s(n) -
r*(n-1)r(n))^2, or, what's better (r(n) - r(n-1)*exp(phi(n))^2 with
phi(n) the (possibly) transmitted phases.

> Pawel

73' Alexander
--
-------------------------------+--------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| technik u. Elektroakustik | Fax : +49-6151-165545
|
| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de
|
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
-------------------------------+--------------------------------------

32. patent doc format

From: Stephen PEARCE <spearce@ccu1.auckland.ac.nz>
Date: Fri, 4 Jul 1997 01:48:45 +1200 (NZT)
To: hfsig@tapr.org
Message-ID: <199707031348.BAA01645@ccu1.auckland.ac.nz>
Rendered body type: text/plain

Hay ...
        What format is this patent document ... does not seem to be
word perfect .. I'm keen to read it

        Thanks
        Stephen

33. RE: [HFSIG:2242] Re: Serial versus parallel, mainly parallel

From: jalocha@axdele.cern.ch
Date: Thu, 3 Jul 1997 17:38:24 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199707031538.RAA13411@dxmint.cern.ch>
Rendered body type: text/plain

>Let "s(n)" be the possibly transmitted signal at time t=nT and "r(n)"
>the received one.
>Usually your metric is (s(n)-CSI*r(n))^2 with CSI the channel state
>information. For differential non-coherent demodulation we take r*(n-1)
>(the last received symbol) as our "CSI", so your metric becomes (s(n) -
>r*(n-1)r(n))^2, or, what's better (r(n) - r(n-1)*exp(phi(n))^2 with
>phi(n) the (possibly) transmitted phases.

Thanks, but I need futher clarifications :-)
- s(n),r(n) are taken as vectors or just amplitudes ?
- CSI is a vector or a number ?
- what is "r*(n-1)" ? Should it be "r(n-1)" ?

Pawel

34. Re: [HFSIG:2243] patent doc format

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Thu, 03 Jul 1997 11:52:40 -0400
To: hfsig@tapr.org
Message-ID: <33BBCAC8.29DE6BED@worldnet.att.net>
Rendered body type: text/plain

Stephen PEARCE wrote:

> Hay ...
>   What format is this patent document ... does not seem to be
> word perfect .. I'm keen to read it
>
>   Thanks
>   Stephen

Aww man! You mean to tell me you have Word Perfect? Well I have Microsoft Word 97... hmm.... how about you format the document to Word Perfect... if that doesn't work then try using it in Note Pad.... ok, I'll talk to ya'll later.

Your Friend,
Patrick McGwier
(Not to be mistaken for his father)

35. Re: [HFSIG:2243] patent doc format

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 03 Jul 1997 08:56:32 -0800
To: hfsig@tapr.org
Message-ID: <199707031554.IAA09313@peak.org>
Rendered body type: text/plain

>Hay ...
> What format is this patent document ... does not seem to be
>word perfect .. I'm keen to read it
>
> Thanks
> Stephen
>
>

I read it perfectly using Microsoft Word 7.0.

--Johan

36. Re: [HFSIG:2244] Re: Serial versus parallel, mainly parallel

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Thu, 03 Jul 1997 18:38:01 +0200
To: hfsig@tapr.org
Message-ID: <33BBD569.3EE71937@zeus.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

jalocha@axdele.cern.ch wrote:

> Thanks, but I need futher clarifications :-)
> - s(n),r(n) are taken as vectors or just amplitudes ?

s(n) and r(n) are complex numbers (baseband representation), i.e. the I and Q part of the signal sampled at nT. I guess that is what you compute by doing the FFT in your OFDM receiver.

> - CSI is a vector or a number ?

CSI is a number (also complex), which you could calculate if you knew the impulse response of the channel

> - what is "r*(n-1)" ? Should it be "r(n-1)" ?
>
> Pawel

the star means complex conjugate(), so z*=(x+jy)*=x-jy

If you want the metric for a sequence of symbols you have to sum all the metrics for all possibly transmitted symbols. Of course you should use the Viterbi algorithm for this.

73' Alexander

--
------------------------------+---------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| technik u. Elektroakustik | Fax : +49-6151-165545
|
| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de
|
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
------------------------------+---------------------------------------

37. Re: [HFSIG:2247] Re: Serial versus parallel, mainly parallel

From: Antony Martin <anthony.martin@gecm.com>
Date: Thu, 03 Jul 1997 22:05:09 +0100
To: hfsig@tapr.org
Message-ID: <970703220509.25667@oas01.gmcl.co.uk>
Rendered body type: text/plain

> jalocha@axdele.cern.ch wrote:
>
> > Thanks, but I need futher clarifications :-)
> > - s(n),r(n) are taken as vectors or just amplitudes ?
>
> s(n) and r(n) are complex numbers (baseband representation), i.e. the I
> and Q part of the signal sampled at nT. I guess that is what you compute
> by doing the FFT in your OFDM receiver.

As you seem to be having difficulty with this, and it appears to be
common knowledge, I'll try and explain:

To obtain a quality indication of a DQPSK signal, you take the phase
of the current symbol and subtract the phase of the previous symbol.
You have to do this anyway to demodulate the DQPSK.

As they're both represented as complex numbers, take the conjugate of
the phase of the previous symbol and complex-multiply it with the
phase of the current symbol.

As only the phase is important, you need to normalize the vectors
to unit magnitude. You will otherwise find the dynamic range a
problem on a fixed-point processor. If you use Q15 maths you may
take 'almost 1' to be unit magnitude to use maximum number range,
but then take care to use correctly saturating arithmetic.

Vectors normalised to unit magnitude are called phasors.

To strip the modulation from this phasor you calculated you must
raise it to the power of 4. ie. complex multiply it by itself 4
times, or square it twice.

If you mave an OFDM system, you should then sum the result
across all tones. Note that this tends to average out noise,
so the result will be most dependent on distortion.

The magnitude of the result is dependent on the demodulation
performance of that symbol, ie. how closely the modulation
on the tones aligns to the 90 degree axes.

The angle of the result is proportional to the frequency
offset present between transmitter and receiver.

If you want a measure dependent on noise, you might try
summing the ABS of imaginary parts. If you've done a
good job correcting the frequency offset it won't
be a significant part of this result.

Normalising is a pain in the arse as it takes more time
than the FFT! If anyone's got any fast normalising
algorithms, I'd love to hear from you!

- --
A.N. Martin   anthony.martin@gecm.com
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

38. Re: [HFSIG:2241] RE: THE MOTHER OF ALL TURDS:K.BERLAND

From: dona@tridelta.com (Don Adams)
Date: Thu, 3 Jul 1997 16:48:35 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199707032048.QAA23400@sparcy.tridelta.com>
Rendered body type: text/plain

>REFERENCE THE 1.4MB mail item.
>
>>Didn't they invent the WWW to PREVENT POSTings like this???
>>
>>1 meg mail messages just SCREW UP my mailer.
>

>I LOST over 20 messages (and about 5 hours) trying to deal with the problems
>created by this message.

I think you should get a better server. Most will allow you to
refuse messages over a size that you specify. This must be
an old system.

39. unsubscribe hfsig

From: keepicon@connect.reach.net (George Elliott)
Date: Fri, 04 Jul 1997 14:28:33 -0400
To: hfsig@tapr.org
Message-ID: <199707041029.KAA05600@connect1.reach.net>
Rendered body type: text/plain

[Empty message body]

40. RE: [HFSIG:2248] Re: Serial versus parallel, mainly parallel

From: jalocha@axdele.cern.ch
Date: Fri, 4 Jul 1997 17:16:48 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199707041516.RAA07724@dxmint.cern.ch>
Rendered body type: text/plain

>> > Thanks, but I need futher clarifications :-)
>> > - s(n),r(n) are taken as vectors or just amplitudes ?
>>
>> s(n) and r(n) are complex numbers (baseband representation), i.e. the I
>> and Q part of the signal sampled at nT. I guess that is what you compute
>> by doing the FFT in your OFDM receiver.
>
>As you seem to be having difficulty with this, and it appears to be
>common knowledge, I'll try and explain:

OK. I understand, but now: Anthony takes the symbol quality
purely from the phase and Alexander uses both phase and amplitude.
I personally am more towards taking the amplitude - for example
an abnormally big signal, although in phase means rather a noise struck
(noise is non-gaussian on HF) than a well received symbol.
What do others think ?

Pawel

41. Re: [HFSIG:2251] Re: Serial versus parallel, mainly parallel

From: Antony Martin <anthony.martin@gecm.com>
Date: Fri, 04 Jul 1997 20:30:04 +0100
To: hfsig@tapr.org
Message-ID: <970704203004.829@oas01.gmcl.co.uk>
Rendered body type: text/plain

> OK. I understand, but now: Anthony takes the symbol quality
> purely from the phase and Alexander uses both phase and amplitude.
> I personally am more towards taking the amplitude - for example
> an abnormally big signal, although in phase means rather a noise struck
> (noise is non-gaussian on HF) than a well received symbol.
> What do others think ?
>
> Pawel
>
I don't think you can infer much from amplitudes in OFDM.

In the presence of multipath, the frequency response across the channel
bandwidth will cause changes in amplitude, as will rapid fading.

An additive interferer equal to the tone amplitude can start to cause
bit errors in a QPSK system. When it causes a bit error, it's phase is
such that the detected amplitude is reduced, not increased.

A reduced tone may be the result of a notch caused by the
frequency response of the channel. If you conclude that this
indicates poorer SNR, you may be wrong, as the noise density
across A HF channel isn't necessarily flat.

If the amplitude of a tone suddenly changes in successive symbols
this may be an indication of an additive impulse which may have
caused an error. But it could also be caused if the code transmitted
created one humongous peak which was heavily clipped.

Nonlinear distortion will also add inter-tone interference which will
affect tone amplitude and phase. Nonlinear distortion includes deliberate
clipping at the modulator and the effects of the transmitter PA.

99.9% of HF noise impulses are of relatively short duration and the
length of symbols in OFDM is much longer. Thus the impulsive nature
of HF noise really doesn't make much difference to OFDM modems.
It's really only of significance to modems 500-5000 baud and the
design of the error control coding used. This is very much indicated
by the 'Longfish' paper reported in the HFsig.

You might like to think of HF noise as random impulsive interference
superimposed on gaussian noise. This kind of interference is present
through much of the RF spectrum. Whether this interference is
a problem depends on your symbol rate. Run at 50Hz and it's averaged
into insignificance. Run at 8MHz on a troposcatter or satellite link
and it's just a little change in DC offset.

Unfortunately, at HF serial modem rates it causes much worse errors
than expected compared to gaussian, so it has to be masked somehow,
usually by FEC. This is OK as impulsive noise really smashes up a few
bits and leaves most bits good, which is what you want for FEC.

An impulse doesn't affect any particular tone anyway; being localised
in time it affects all tones equally. So comparing one tone with the
others to find a spike won't help. In this respect gaussian noise
affects all tones randomly, but an impulse has a correlated effect
across the tones. This may be an interesting aspect to persue.

I don't use quality information to weight bits for error correction.
I use it to keep the timing right and to keep the frequency offset
correction right. If these aren't right, you will get errors, lots of
them! It's also used in determination of 'signal lost'.

It might be interesting to re-plot my modem BER vs SNR curves with off-air
noise instead of simulated gaussian to see if it makes any difference.

Bi Bi, Anthony

42. Re: [HFSIG:2236] HFSIG digest 350

From: RedHaines@centuryinter.net
Date: Fri, 04 Jul 1997 16:42:47 -0600
To: "by way of The Whiting Household <hfsig@tapr.org>" <hfsig@tapr.org>
Message-ID: <33BD7C67.517@centuryinter.net>
Rendered body type: text/plain

Hi, Rick;

Is this another of those addresses that goes to everyone? Oh, well.

I just set up a station in the new house, and was trying out ham and
internet simultaneously when I received your message. No, I haven't
read the book!

I put up a Hi-Gain trapped vertical antenna, adjusted for the CW portion
of the bands. May or may not be able to join folks on 20M SSB. I'm
flying the tow plane this Saturday, so won't find out this week unless
the wx is bad for soaring!

I trust the job continues to be a pleasant challange. How's school?

I'll try to give you a shout next time I come to MPLS.

73 de Red

43. unsubscribe hfsig

From: Ed Lyons <eel@ipass.net>
Date: Sat, 05 Jul 1997 15:39:41 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970705193941.006951d8@ipass.net>
Rendered body type: text/plain

[Empty message body]

44. [HFSIG:2254] unsubscribe hfsig

From: RAYMOND CZYZEWSKI <wa2sei@nac.net>
Date: Sat, 05 Jul 1997 18:32:00 -0400

To: hfsig@tapr.org
Message-ID: <33BECB60.72D4@nac.net>
Rendered body type: text/plain

[Empty message body]

45. [HFSIG:2255] unsubscribe hfsig

From: Monte.Tremont@nextel.com (Monte Tremont)
Date: Tue, 8 Jul 1997 09:34:34 -0400
To: hfsig@tapr.org
Message-ID: <00016615.@nextel.com>
Rendered body type: text/plain

[Empty message body]

46. unsubscribe hfsig

From: Bill Carver <bcarver@magiclink.com>
Date: Tue, 8 Jul 1997 17:34:21 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <199707082234.RAA05451@tapr.org>
Rendered body type: text/plain

[Empty message body]

47. unsubscribe hfsig

From: Ed Lyons <eel@ipass.net>
Date: Tue, 08 Jul 1997 20:27:23 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970709002723.0068fec8@ipass.net>
Rendered body type: text/plain

[Empty message body]

48. BPSK & Parallel modem tests status:

From: Eduardo Jacob <edu@kender.es>
Date: Wed, 09 Jul 1997 07:36:11 -0100
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19970709073611.006b8028@kender.es>
Rendered body type: text/plain

9At 10:46 1/07/97 -0500, you wrote:

>1) With a group of amateurs monitoring in the UK and the Netherlands using
>sensitive narrow-band signal detection software, the KC7WW beacon that
>operated on 14070 (carrier) was not detected. I guess if there is no
>propagation, there is no signal to work with, regardless of what you do. The
>experiment will now reverse direction with PA0IHD transmitting a beacon for
>a week on 14070, so this is an opportunity to see if we succeed this time.

I have been trying to hear the beacon, but in that precise frequency there is much pactor PMS activity and ocasional pactor-QSO. The antenna I use isn't probably the best (a Cushcraft R7 vertical) but is the only one I could put in the building were I live.

As I don't know the precise characteristics of BPSK I don't know if it should pass through the pactor signals...

BEst regards

Eduardo

PD: Another frecuency to try?, perhaps on 18 Mhz, perhaps in a CW portion of the 14 Mhz (the QSO are not as frecuency fixed)?

49. [HFSIG:2257] unsubscribe hfsig

From: Monte.Tremont@nextel.com (Monte Tremont)
Date: Wed, 9 Jul 1997 08:27:58 -0400
To: hfsig@tapr.org
Message-ID: <000184E5.@nextel.com>
Rendered body type: text/plain

[Empty message body]

50. Re: [HFSIG:2260] unsubscribe hfsig

From: dona@tridelta.com (Don Adams)
Date: Wed, 9 Jul 1997 09:43:58 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199707091343.JAA13221@sparcy.tridelta.com>

Rendered body type: text/plain

No offense, but I understand it's vacation time and
a lot of people want to unscubribe, but don't you have to
send it to the listserver@tapr with unsubsribe hfsig in the
body (or something like that) and not to the hfsig (or
does this work?)

>
>
>
>
>

51. Re: [HFSIG:2202] Re: Serial versus parallel, mainly parallel

From: Debbie Douglass <douglass@texas.net>
Date: Wed, 09 Jul 1997 18:44:04 -0500
To: hfsig@tapr.org
Message-ID: <3.0.1.32.19970709184404.006b0f80@texas.net>
Rendered body type: text/plain

I am not the person you intended to send this mail to. E-mail address
debbie@texas.net is not for the person you intended to mail to. Please
check for the current address.

At 11:06 AM 6/30/97 -0500, you wrote:
>Alexander:
>>Constant envelope on each carrier of an OFDM transmission is useless as
>>the overall signal is of course not constant envelope. So this can't be
>>the reason for using SQPSK.
>
>I think that in a multitone transmition the non-constant envelopes
>of the tones do matter because the overall signal will get this envelope
>(you hear brrrrr. at the symbol rate) which is not good for the peak/RMS
>ratio.
>
>>Furthemore SQPSK (or OQPSK) can't be easily
>>demodulated non-coherently.
>
>I demodulate non-coherently anyway - I only look at the phase difference,
>never at the absolute phase.
>
>I thought that maybe the SQPSK is to get the soft decisions easier ?
>
>How do you get the soft decisions anyway in differential QPSK ?
>Should the weight be proportional to the correlation I get by
>multiplying two succesive vectors ?
>But I can get very strong (=good ?) correlations for tones jammed
>by CW carriers ?
>
>Pawel
>
>
>
>

52. Re: [HFSIG:2195] Re: Serial versus parallel

From: Debbie Douglass <douglass@texas.net>
Date: Wed, 09 Jul 1997 18:40:16 -0500
To: hfsig@tapr.org
Message-ID: <3.0.1.32.19970709184016.006acfe0@texas.net>
Rendered body type: text/plain

I am not the person you intended to send this mail to. E-mail address
debbie@texas.net is not the address for the person you thought it was.
Please check for the correct address.

At 08:38 AM 6/29/97 -0500, you wrote:
>>Do you also advocate the use of guard bands in the OFDM demodulation
>>process, i.e., instead of using all the data from the entire symbol
>>time, the FFT process only uses a slice out of the middle of the baud,
>>thus leaving a guard band each side to absorb multipath jitter?
>
>Note this can be accomplished with a window function on each pulse.

>You can use any of the traditional windowing functions: Hamming, Hanning,
>Blackman, Kaiser, etc.
>
>>Continuing on the issue of OFDM; Some of us have been playing around with
>>RC pulse shaping applied to individual carriers within the OFDM ensamble,
>>only to find that one then needs wider spacing the 1/T orthogonal
spacing. The
>
>Of course! This is exactly the same phenomenon as a windowing function
>applied to the input of a FFT knocking down the sidelobe leakage at
>the cost of broadening the main lobe. Pick your poison; you can't have
>both a narrow main lobe and low sidelobes.
>
>Phil
>
>
>
>
>

53. Re: [HFSIG:2248] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 10 Jul 1997 20:19:10 -0700 (PDT)
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <199707110319.UAA28455@servo.qualcomm.com>
Rendered body type: text/plain

>As only the phase is important, you need to normalize the vectors
>to unit magnitude. You will otherwise find the dynamic range a
>problem on a fixed-point processor. If you use Q15 maths you may
>take 'almost 1' to be unit magnitude to use maximum number range,
>but then take care to use correctly saturating arithmetic.

The magnitudes are important if you're doing soft-decision Viterbi
decoding.

Re normalization and dynamic range, that's what floating point
processors are for. The Pentium's FPU is much faster than most
so-called integer "DSP" chips...

Phil

54. [arvydas.vaisnys@jpl.nasa.gov: digital broadcasting]

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 10 Jul 1997 20:44:28 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707110344.UAA28635@servo.qualcomm.com>
Rendered body type: text/plain

I'm forwarding this to hfsig with the author's permission. He's not
on this list, so contact him directly. --Phil

------- Start of forwarded message -------
Date: Tue, 24 Jun 97 14:52:41 PDT
X-Sender: vaisnys@137.78.161.31
X-Mailer: Windows Eudora Pro Version 3.0 (16)
To: karn
From: Arvydas Vaisnys <arvydas.vaisnys@jpl.nasa.gov>
Subject: digital broadcasting
Mime-Version: 1.0
Content-Type: text/plain; charset="us-ascii"

Phil,

We have been working on the development of a short wave digital audio
broadcasting system for the VOA and have a modem working in software on a
pentium. In non-real time unfortunately. It requires the recording of a low
IF on a DAT, then inputing the resulting data stream into the computer for
processing. Thanks to some heavy duty equalization we achieved some good
test results: this May transmitted 16kbps compressed audio using 8PSK at
12kW from Delano to Washington DC at 18 MHz. The test segment included some
DSB AM and the comparison is dramatic. We hope to achieve higher data rates
by using even higher orders of modulation.

We think a useful step in this development could be to do field trials and
have some radio amateurs with receivers evaluate system performance. The
catch is the availability of receivers. The VOA being a government
organization has little money. Do you think some radio amateur group might

be interested in a joint development of an inexpensive receiver? This could
be something PC circuit card based that could accomplish some of the
processing, then let the computer do the rest. We would be happy to share
our software.

Consider this a trial balloon. I don't know how many radio amateurs are
interested in broadcasting, but the potential for this service is real.
International broadcasters have started a push for conversion to digital
and there are other system development efforts in Germany and France.
Please let me know if you have any ideas.

Arv
------- End of forwarded message -------

55. Re: [HFSIG:2251] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 10 Jul 1997 21:46:15 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707110446.VAA29239@servo.qualcomm.com>
Rendered body type: text/plain

>OK. I understand, but now: Anthony takes the symbol quality
>purely from the phase and Alexander uses both phase and amplitude.
>I personally am more towards taking the amplitude - for example
>an abnormally big signal, although in phase means rather a noise struck
>(noise is non-gaussian on HF) than a well received symbol.
>What do others think ?

The correct procedure for demodulating differential BPSK to form soft
decision samples (e.g., for Viterbi decoding) is to take the complex
dot product of the current symbol with the previous symbol. Two strong
consecutive symbols of the same phase will give a large positive
sample, and two strong consecutive symbols of opposite phase will give
a large negative sample.

And if you take the cross product of the two complex symbols, you'll
get your AFC error signal. (Although differential BPSK doesn't require
carrier phase recovery, it does require an approximate estimate of
carrier frequency.)

Phil

56. Re: [HFSIG:2264] Re: Serial versus parallel, mainly parallel

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Fri, 11 Jul 1997 10:29:49 +0200
To: hfsig@tapr.org
Message-ID: <33C5EEFD.5956@ife.ee.ethz.ch>
Rendered body type: text/plain

> Re normalization and dynamic range, that's what floating point
> processors are for. The Pentium's FPU is much faster than most
> so-called integer "DSP" chips...

"Much" is kind of an exaggeration... (well, much is probably correct
if you compare the pentium with a sub 1$ DSP :-))

Modern floating point DSPs give you about the same performance
as the Pentium for less than 20$, and that already including
some fast RAM, and consuming a fraction of the power than your
P5.

The advantage of using a P5 is that it is simpler to develop
the applications (once you know how to work around the OS to
do soft realtime stuff and you know how to drive soundcards; Linux
is in this respect much simpler than M$ stuff) than on the DSP,
simply because the tools are more mature.

The above is the experience gained from implementing just about
every major amateur digital mode on both a P5 and at least one
DSP :-)

Tom

57. Re: [HFSIG:2264] Re: Serial versus parallel, mainly parallel

From: Antony Martin <anthony.martin@gecm.com>
Date: Fri, 11 Jul 1997 20:30:58 +0100
To: hfsig@tapr.org
Message-ID: <970711203058.23756@oas01.gmcl.co.uk>

Rendered body type: text/plain

> >As only the phase is important, you need to normalize the vectors
> >to unit magnitude. You will otherwise find the dynamic range a
> >problem on a fixed-point processor. If you use Q15 maths you may
> >take 'almost 1' to be unit magnitude to use maximum number range,
> >but then take care to use correctly saturating arithmetic.
>
> The magnitudes are important if you're doing soft-decision Viterbi
> decoding.
>
> Re normalization and dynamic range, that's what floating point
> processors are for. The Pentium's FPU is much faster than most
> so-called integer "DSP" chips...
>
> Phil
>

Yes but my DSP costs under 30 UKP (about 50US$) and includes all memory
and interfaces necessary. It takes 23mA from a 3V supply when fully loaded,
considerably less on a parallel modem algorithm. An entire modem with data
converters takes less than 40mA.

We have to support products for 15 years, but the pentium chip & it's
necessary cache controllers, i/o megachips and bus support devices, heatsinks
special 3V regulators etcetera etcetera will be obsolete & unobtainable next
year. Not to mention the EMC problems.

When size and power dissipation are not a problem, go for a floating-point
DSP. I would. If they are a problem, then you need fixed-point. TMS320C32's
are very cheap, but you need external memory, and fast RAM cranks up the cost.
The new cut-down Sharc is also lowish cost and includes a lot of fast on-chip
RAM.

If you're comparing processors, floating-point is about 3 times faster
on a SPARC than on a pentium p166. Intel traded FPU die area for Cache size
and hence the pentium p166 has similar float performance to a 486DX66. These
comparisons were obtained from the freely-available C-source for LPC-10
speech coding when compiled and run on various platforms. On the
pentium the algorithm would not run real-time. On a SPARC it ran about
30% faster than real-time. On a fixed-point DSP a conversion of it
runs real-time (just) with some assembler coded routines. The same
integer conversion of it runs much faster than real time on a PC.

A lot of modem algoritms could be readily implemented directly on a
pentium PC. Sample I/O is the problem as you have to work with a large
number of different cards, and interrupt handling is very inefficient.
To do it under windows you would have to write it as a .DRV, as windows
won't guarantee your program will keep up with sample I/O when other
things are happening. In unix, sample i/o is easy as the /dev/audio does
all the work.

Of those out there that have a PC sound card, how many could tell me
if the hardware can do 8kHz sample input and output (rather than what
the software supplied with it allows you to do)? What accuracy sample
rate does it guarantee? Can it do in & out simultaneously? You'll be
suprised how little documentation you've actually got. And you would
be amazed how many variants of soundblaster there are.

If you have a decent unix system, type "man 4 audio" and you'll get
about 10 pages on how to program audio i/o.

Anthony

58. Re: [HFSIG:2268] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 11 Jul 1997 16:14:12 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707112314.QAA08119@servo.qualcomm.com>
Rendered body type: text/plain

>Yes but my DSP costs under 30 UKP (about 50US$) and includes all memory
>and interfaces necessary. It takes 23mA from a 3V supply when fully loaded,
>considerably less on a parallel modem algorithm. An entire modem with data
>converters takes less than 40mA.

And your modem plugs into just what kind of terminal equipment...?
Unless it's part of a standalone portable digital voice system, it's
most likely plugged into a PC! And today that PC most likely contains
a Pentium that twiddles its thumbs for several million cycles waiting

for each bit of data to come in on its serial port from your modem.

>We have to support products for 15 years, but the pentium chip & it's
>necessary cache controllers, i/o megachips and bus support devices, heatsinks
>special 3V regulators etcetera etcetera will be obsolete & unobtainable next
>year. Not to mention the EMC problems.

Today's Pentium chip may well be obsolete next year, but it'll be
replaced by the Pentium Pro or the Pentium II, which will in turn be
replaced by something newer with even more execution pipelines and an
even higher clock speed. But it will still run every x86 program
written since the dawn of time (1979, when the original 8086 was
released). If anything, the Intel x86 processor line has been *too*
focused on absolute object code backward compatibility. This has
required some really heroic chip design to make this rather limited
programming model perform decently fast. The best example are the
shadow registers introduced in the P6 (Pentium Pro) to overcome the
severe register shortage of the Intel model that makes it otherwise
very difficult to keep multiple superscalar pipelines busy.

The Power PC is a much better chip than the Pentium, thanks largely to
its much bigger register array. You don't have to turn handsprings in
custom assembler to keep its pipelines full, something I have a lot of
experience doing on the Pentium. (See my web pages for pipelined FIR
and FFT asm code). Compilers for decent RISC machines already know how
to do an adequate job.

But the Power PC isn't backward compatible with all that legacy Intel
software, which is why it has been largely a failure in the market.
The Pentium is everywhere, like it or not, so that's the machine I use.
It's still roughly equivalent to a Cray 1 in floating point performance,
and that's hard to ignore.

>and hence the pentium p166 has similar float performance to a 486DX66. These

This is simply not true. The Pentium's FPU, properly programmed, can
turn out one instruction per clock, including multiplies. (My Pentium
FIR asm code does a tap in 3 clocks, floating point.) Even if the
Pentium had the same FPU design as the 486, the higher clock rate
alone would make a substantial difference.

I certainly won't argue that the SPARC isn't a better
architecture. I'm typing on one right now here in the office. But the
machines at home (the ones I paid for myself) are all Pentiums.

>number of different cards, and interrupt handling is very inefficient.

Again I won't defend the PC architecture, but most sound cards now use
double-buffered DMA with interrupts occurring at the halfway and wraparound
points. So the interrupt rate isn't bad, even at high sample rates.

>To do it under windows you would have to write it as a .DRV, as windows

So don't use Windows, use Linux. Or, if you are willing to dedicate a
machine to a real-time task, use DOS and a C environment with a
protected-mode extender like DJGPP (free code).

The bottom line is that hams interested in experimenting with DSP are
still much better off using standard PCs and sound cards and readily
available (often for free) development environments than in buying
overpriced, one-of-a-kind DSP development boards.

Phil

59. Re: [HFSIG:2266] Re: Serial versus parallel, mainly parallel

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Fri, 11 Jul 1997 06:57:48 -0400
To: hfsig@tapr.org
Message-ID: <33C611AC.FE04918A@worldnet.att.net>
Rendered body type: text/plain

Phil Karn wrote:

> The correct procedure for demodulating differential BPSK to form soft
> decision samples (e.g., for Viterbi decoding) is to take the complex
> dot product of the current symbol with the previous symbol. Two strong
>
> consecutive symbols of the same phase will give a large positive
> sample, and two strong consecutive symbols of opposite phase will give

>
> a large negative sample.
>
> And if you take the cross product of the two complex symbols, you'll
> get your AFC error signal. (Although differential BPSK doesn't require
>
> carrier phase recovery, it does require an approximate estimate of
> carrier frequency.)
>
> Phil

You need to make clear that "is negative" with a complex number means
that the real part is what you
are talking about and that the imaginary part is the AFC error signal,
i.e. if the real part of the
product is the symbol detector and the imaginary part is the "AFC"
signal. The AFC signals needs
to be low passed filtered to be of use.

Bob

--
****

* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd.   * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520   * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963   * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061   * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240   * Council. Brownie Troop 193
*
****

60. Re: [HFSIG:2270] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>

Date: Fri, 11 Jul 1997 19:13:55 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707120213.TAA10091@servo.qualcomm.com>
Rendered body type: text/plain

>You need to make clear that "is negative" with a complex number means
>that the real part is what you are talking about and that the
>imaginary part is the AFC error signal,

Well, since the dot product of two complex numbers is a simple scalar,
the resulting demodulated symbol is inherently real.

You're quite right that the AFC signal derived from the cross product
must be filtered. Turns out you can usually use a pretty tight filter
here, much tighter than the loop filter you could use in a coherent
carrier recovery loop (e.g., a Costas loop). That's because while a
Rayleigh fading signal can have very rapid phase fluctuations that
defeat narrow PLL loop filters, its frequency remains relatively
stable unless there is also gross doppler shift in the channel.

Phil

61. Re: [HFSIG:2268] Re: Serial versus parallel, mainly parallel

From: Thomas Sailer <sailer@ife.ee.ethz.ch>

Date: Mon, 14 Jul 1997 09:58:04 +0200
To: hfsig@tapr.org
Message-ID: <33C9DC0C.CA5@ife.ee.ethz.ch>
Rendered body type: text/plain

Antony Martin wrote:

> on a SPARC than on a pentium p166. Intel traded FPU die area for Cache size
> and hence the pentium p166 has similar float performance to a 486DX66. These

That can't be. My comparisons between a 486DX2/66 and a P75 (both
Intel) resulted in both CPU's having about the same
integer performance, while FPU performance of the P75 was roughly 3

times faster than the 486.

> number of different cards, and interrupt handling is very inefficient.
> To do it under windows you would have to write it as a .DRV, as windows
> won't guarantee your program will keep up with sample I/O when other

Nope, won't work that way. .DRV's are just .DLL renamed, read they
execute also in ring 3. To do it properly you need to write
a VXD (and Phil: to write a VxD, you'll need to do some ASM too :-))
See http://home.pages.de/~flexnet for some packet radio modems
using a VXD.

> the software supplied with it allows you to do)? What accuracy sample
> rate does it guarantee? Can it do in & out simultaneously? You'll be

Accuracy is definitely a problem, some soundcards use ceramic
resonnators with maybe 1% deviation from their nominal frequency...
And quality isn't even correlated to price, I've seen good
and very cheap cards, and expensive and very bad ones

> suprised how little documentation you've actually got. And you would
> be amazed how many variants of soundblaster there are.

Another problem is frequency response. Esp. Creative cards
have an analog "mixer chip" in the signal path which has
bass and treble filters which introduce phase shifts.

Tom

62. Re: [HFSIG:2269] Re: Serial versus parallel, mainly parallel

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Mon, 14 Jul 1997 10:37:58 +0200
To: hfsig@tapr.org
Message-ID: <33C9E566.38E5@ife.ee.ethz.ch>
Rendered body type: text/plain

> And your modem plugs into just what kind of terminal equipment...?
> Unless it's part of a standalone portable digital voice system, it's
> most likely plugged into a PC! And today that PC most likely contains
> a Pentium that twiddles its thumbs for several million cycles waiting
> for each bit of data to come in on its serial port from your modem.

Well and what are you going to do if the operator gets bored
of the HF circuit and wants to play quake until the message
arrives at the other end? Just kidding :-)

> Again I won't defend the PC architecture, but most sound cards now use
> double-buffered DMA with interrupts occurring at the halfway and wraparound
> points. So the interrupt rate isn't bad, even at high sample rates.

This won't solve the latency problem! Synchronous modes such
as those used on HF, you just cannot wait 100ms for the interrupt to
start the processing! Also on VHF packet, CSMA channel access
degrades rapidly if your DCD samples date back too long...

> So don't use Windows, use Linux. Or, if you are willing to dedicate a

If you want to experiment, you're fine with Linux. If you want
to reach mainstream status, you'll have to do windoze 95.
An example is ham TCP/IP. While your net package spurred
some experimentation with TCP/IP, it reached mainstream status
here only after an easy to install TCP/IP package for Windoze 95
was available lately (FlexIP)

Tom

63. Re: [HFSIG:2273] Re: Serial versus parallel, mainly parallel

From: Antony Martin <anthony.martin@gecm.com>
Date: Mon, 14 Jul 1997 13:15:59 +0100
To: hfsig@tapr.org
Message-ID: <970714131559.4003@oas01.gmcl.co.uk>
Rendered body type: text/plain

> > And your modem plugs into just what kind of terminal equipment...?
> > Unless it's part of a standalone portable digital voice system, it's
> > most likely plugged into a PC! And today that PC most likely contains
> > a Pentium that twiddles its thumbs for several million cycles waiting
> > for each bit of data to come in on its serial port from your modem.
>

> Well and what are you going to do if the operator gets bored
> of the HF circuit and wants to play quake until the message
> arrives at the other end? Just kidding :-)
>
Whatever the pros and cons of processor performance, I'm inclined to
agree that the PC is an appropriate platform for many amateurs to
experiment with digits on HF. The fact that the number of variants
of hardware must equal the number of amateurs must make it difficult
to ensure software will work on PCs other than your own.

But it isn't the cheapest by a long street - A PC will cost you 1000
UKP but you could buy a TMS320C50 DSK for 95 UKP and a character
terminal from a computer auction for 5 UKP. Replace the EPROM in the DSK
and it's an HF modem. You can be on air with digits for only 100 UKP.

The problem with HF comms in commercial applications is that they
have a habit of using it in most remote inhospitable places. If it
isn't in a desert, or in the arctic, it's up a mountain somewhere
remote from cities where reception is unobstructed and interference is
low. Or it's bouncing around the back of a landrover on a dirt road in
Bosnia, or crashing thru the north sea. This is understandable as in
most places people find it easier to pick up a telephone. Amateurs,
though, need only one excuse to use HF: a licence.

Unfortunately PCs don't last very long when dropped 1.2 metres onto
concrete and submerged 1 metre underwater for 4 hours... They don't
usually start at -40 degrees and something's usually gone bang before
+50 degrees. PCs MTBF (Mean time between failures) isn't high enough
for unattended remote operation, far too many transistors and
mechanical whirry things, the silicon runs too hot, and not enough screws.

Except possibly for remote farms in Australia or South Africa, most
commercial applications of HF can't use PCs.

But if you want to experiment using your PC as a washing machine
controller, an engine management controller, and antilock brake system,
it's fine by me. You don't have to use it for wordprocessing
internet and multimedia...

Anthony.

64. Re: [HFSIG:2273] Re: Serial versus parallel, mainly parallel

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Mon, 14 Jul 1997 08:27:04 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.970714082204.11033F-100000@www.kelly-afb.org>
Rendered body type: text/plain

Tom,

On Mon, 14 Jul 1997, Thomas Sailer wrote:

>
> > So don't use Windows, use Linux. Or, if you are willing to dedicate a
>
> If you want to experiment, you're fine with Linux. If you want
> to reach mainstream status, you'll have to do windoze 95.
> An example is ham TCP/IP. While your net package spurred
> some experimentation with TCP/IP, it reached mainstream status
> here only after an easy to install TCP/IP package for Windoze 95
> was available lately (FlexIP)
>
> Tom
>
>

I liken MicroSquat Windoze, 95 and below, to the 27 MHz Citizens Band
and Linux to Amateur Radio.

Its just where you wanna play...with the CBers or Hams.

Walt/K5YFW

```
    ========================================================================
    |        If I wanted "Windows", I'd knock a hole in my wall.           |
    ========================================================================
    |                                  |                                   |
    |                                  | The Micro$oft operating system    |
```

```
| Walt DuBose - K5YFW               | didn't get as bad as it is      |
| E-Mail k5yfw@www.kelly-afb.org    | overnight.  I has taken over 10 |
| Business Telephone: (210)925-1171 | years of careful, calculated,   |
|     Home Telephone: (210)696-3196 | development.                    |
|                                   |                                 |
=====================================================================
```

65. Re: [HFSIG:2267] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 14 Jul 1997 21:13:30 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707150413.VAA17622@servo.qualcomm.com>
Rendered body type: text/plain

>Modern floating point DSPs give you about the same performance
>as the Pentium for less than 20$, and that already including
>some fast RAM, and consuming a fraction of the power than your
>P5.

Yeah, and as I said last week, chances are very good that you're going to plug your $20 DSP into the serial port of a Pentium that will twiddle its thumbs for a few million clocks between each bit from the DSP!

I do not doubt the value of DSPs in mass scale production of a dedicated application where size, weight and power are at a premium, and a general purpose computer isn't an integral part of the system. That perfectly describes our cell phones, which we've turned out in sufficient numbers to give several to every ham in the US.

But for small-scale ham experimentation and self-education, I could never for the life of me understand the ham (TAPR?) inclination to spend considerable resources designing, building and replicating special-purpose hardware before the desired functionality has even been prototyped and tested on a general purpose computer and an optimized system design done based on this experience.

Especially since the hams most likely to be interested in the special-purpose box probably already have a recent PC and would be quite willing to use it for the application in question, at least on an experimental basis, before blowing money on specialized hardware.

It's putting the cart well before the horse. Or in my business, we call it "letting hardware engineers do system engineering". :-)

Phil

66. Re: [HFSIG:2276] Re: Serial versus parallel, mainly parallel

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Tue, 15 Jul 1997 10:02:07 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199707150702.KAA00715@boojum.hut.fi>
Rendered body type: text/plain

> Especially since the hams most likely to be interested in the
> special-purpose box probably already have a recent PC and would be
> quite willing to use it for the application in question, at least on
> an experimental basis, before blowing money on specialized hardware.
>
> It's putting the cart well before the horse. Or in my business, we
> call it "letting hardware engineers do system engineering". :-)

Ok, the first demodulator I made was a Ge diode :-). We have really come a long way from that, but I still prefer the "bottom-up" approach from the "top-down". I feel much more confortable with a simple hardware with nothing extra that might make things complicated. This may well be due to my ignorance. So Phil, could you give us pointers to the necessary know-how on doing DSP in a Pentium? If you have time a simple HOWTO on "DSP modems with a PC" would be very valuable. I recently purchased a Linux box to home and would preferably do all work on that.

Jaakko/OH7BF

67. Re: [HFSIG:2273] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 15 Jul 1997 00:07:15 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707150707.AAA19241@servo.qualcomm.com>
Rendered body type: text/plain

>This won't solve the latency problem! Synchronous modes such
>as those used on HF, you just cannot wait 100ms for the interrupt to
>start the processing! Also on VHF packet, CSMA channel access
>degrades rapidly if your DCD samples date back too long...

I didn't say soundcard automatic DMA would solve the latency problem,
only that it would reduce or eliminate data loss caused by CPU
interrupt latency.

But speaking of latency, the real problem is the use of such tight
synchronicity in the first place. Who says I should design my schemes
to sound like AMTOR? Why can't I make long transmissions with long
interleaver spans to combat fading, maximize coding gains, minimize
wasted channel time due to prop delays *and* save my transceiver T/R
relays while I'm at it?

Good point about CSMA latencies. But I don't recommend sound cards for
the very high speed modems we should be running in local area VHF/UHF
networks. They're more appropriate for weak signal links where you
can fit everything in audio-like bandwidths without having to resort
to bandwidth-limited modulation schemes.

Phil

68. Re: [HFSIG:2274] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 15 Jul 1997 00:16:46 -0700 (PDT)
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <199707150716.AAA19303@servo.qualcomm.com>
Rendered body type: text/plain

>Whatever the pros and cons of processor performance, I'm inclined to
>agree that the PC is an appropriate platform for many amateurs to
>experiment with digits on HF. The fact that the number of variants

Agreed.

>of hardware must equal the number of amateurs must make it difficult
>to ensure software will work on PCs other than your own.

Only in the UK, where they have all sorts of strange things they call
"personal computers" that seem to be named after the things that fall
off of oak trees. :-)

Seriously, the world does seem to be rapidly converging, for better or
worse, on personal computers using Intel microprocessors. The operating
systems may vary, but programs written for one have been known to be
ported successfully to others.

>But it isn't the cheapest by a long street - A PC will cost you 1000
>UKP but you could buy a TMS320C50 DSK for 95 UKP and a character
>terminal from a computer auction for 5 UKP. Replace the EPROM in the DSK
>and it's an HF modem. You can be on air with digits for only 100 UKP.

This assumes you don't already own a PC. And it assumes that somebody
else develops the necessary firmware for your precise hardware. And
that you don't care to experiment with it yourself, lest you have to
go out and assemble all the necessary support tools.

>Unfortunately PCs don't last very long when dropped 1.2 metres onto
>concrete and submerged 1 metre underwater for 4 hours... They don't

This may be true, but I keep asking -- what good is an HF modem
without a computer of some sort to plug into it? I suppose you may be
lucky enough to have a HF modem fast enough to support digital voice,
in which case you could make a nice portable HF digital voice radio.

I suspect that the vast majority of hams into the HF digital modes are
interested in sending textual information, because it makes much
better use of the limited data rates available on HF, and most have
already replaced their ancient dumb terminals with computers that
allow offline editing and storage.

Phil

69. Re: [HFSIG:2274] Re: Serial versus parallel, mainly parallel

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Tue, 15 Jul 1997 11:42:37 +0200
To: hfsig@tapr.org
Message-ID: <33CB460D.7127@ife.ee.ethz.ch>
Rendered body type: text/plain

> Unfortunately PCs don't last very long when dropped 1.2 metres onto
> concrete and submerged 1 metre underwater for 4 hours... They don't
> usually start at -40 degrees and something's usually gone bang before
> +50 degrees. PCs MTBF (Mean time between failures) isn't high enough
> for unattended remote operation, far too many transistors and
> mechanical whirry things, the silicon runs too hot, and not enough screws.

Valid point. That's why we don't use PC's in packet radio node
controllers, which are often installed in harsh environments, such
as mountain tops.

I heard stories of people travelling a few hours just to
hit "F1" so that the PC could reboot :-)

Tom

70. Re: Serial versus parallel, mainly parallel

From: "Bergzen Hakan, KR" <hakan.bergzen@enator.se>
Date: Tue, 15 Jul 1997 12:06:00 +0100
To: hfsig <hfsig@tapr.org>
Message-ID: <33D0D825@noak>
Rendered body type: text/plain

Hello,

I hope I am excused for jumping into the ongoing title-deviated discussion
like this :-)

When it comes to PC versus DSP implementation I think the cost issue has
been somewhat over-rated. More or less every ham has a PC (at least those on
this newsgroup) and could also very well afford the DSP-kit implementation.

In my humble opinion it is more a matter of what you as a ham are interested
in experimenting with.

Personally, I would not like to use Linux since all my other programs (for
good or bad) are housed in the MS-Windows environment, and a temporary
switch would make my PC just as specialised as a DSP board.

Even though I believe another motive behind going for a dedicated DSP board
(it is much easier for a company to put a higher price on something physical
that you can actually see, than just software), one always seems to come up
with some sort of black box. E.g. the soundblaster i/o will probably need
extrnal circuitry to handle impedance matching and keyline signalling.

However I just recently heard of a complete Mil-Std-188-110A serial tone
mode modem implementation in a Pentium PC, so it most certainly has the
horse-power to do the work required for the modem schemes thought of here.

The two methods have actually more similarities than differencies, ie they
are both based on standardised platforms (EVM starter kits are quite
standardised really) and the freely available software/hardware gives a ham
a possibility to learn modem functionality and implementation (as opposed to
buying a PTC/Clover/something other modem). In short, they both take back
the development initiative from the ham radio companies to the original ham
operators.

Hakan

71. Re: [HFSIG:2277] Re: Serial versus parallel, mainly parallel

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Tue, 15 Jul 1997 12:34:08 +0200
To: hfsig@tapr.org
Message-ID: <33CB5220.501A@ife.ee.ethz.ch>
Rendered body type: text/plain

> due to my ignorance. So Phil, could you give us pointers to the necessary
> know-how on doing DSP in a Pentium? If you have time a simple HOWTO on
> "DSP modems with a PC" would be very valuable. I recently purchased
> a Linux box to home and would preferably do all work on that.

It's easy. Just grab C compiler and read the appropriate
"HOWTO". From my experience, the overall gain won't be big

if you hack anything in asm...

If you want to know how to do audio IO under Linux, read the
manual at http://www.4front-tech.com/pguide/, if you're interested
in Windoze, read the "MMSYSTEM" docs from M$, especially
the WaveIn* and WaveOut* functions. For DOS, search the Web
for Ethan Brodsky's soundblaster page...

73s
Tom

72. Re: [HFSIG:2276] Re: Serial versus parallel, mainly parallel

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Tue, 15 Jul 1997 12:45:15 +0200
To: hfsig@tapr.org
Message-ID: <33CB54BB.69CD@ife.ee.ethz.ch>
Rendered body type: text/plain

> Yeah, and as I said last week, chances are very good that you're going
> to plug your $20 DSP into the serial port of a Pentium that will twiddle
> its thumbs for a few million clocks between each bit from the DSP!

Install Windoze 95 and you'll see that your expensive CPU is
busy trashing segment register caches :-)

Another point: If you have a DSP dedicated to your job, you can
control the software environment completely. If you want to
use the P5, there's an enormous amount of different configurations,
eg. Windoze 95, 95a, 95b, OSR2, or Linux 2.x.y where x=0,1 and
y=0..40 :-) My stuff being mainly amateur, I don't quite care
much as long as it runs on my configuration, but if I'd do
a commercial product, I probably cannot live with this.
If you don't believe me, I'd be glad to forward you those
5 to 10 messages daily asking about how to install the stuff...

> But for small-scale ham experimentation and self-education, I could
> never for the life of me understand the ham (TAPR?) inclination to
> spend considerable resources designing, building and replicating

That's something I didn't understand too... There are plenty of
cheap platforms around to experiment with, so why build
yet another one?

> quite willing to use it for the application in question, at least on
> an experimental basis, before blowing money on specialized hardware.

IMNSHO the average ham doesn't understand much of it, so he's thinking
that a box that costed 1000$ must perform better than free
software... the same argument basically as why companies use
NT instead of Linux :-)

Tom
--
Thomas (Tom) Sailer   EMail: sailer@ife.ee.ethz.ch
Weinbergstrasse 76   Ham Radio: hb9jnx @ hb9w.che.eu
CH-8408 Winterthur   Phone: ++41 52 222 32 81

73. Re: [HFSIG:2278] Re: Serial versus parallel, mainly parallel

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Tue, 15 Jul 1997 12:50:31 +0200
To: hfsig@tapr.org
Message-ID: <33CB55F7.2719@ife.ee.ethz.ch>
Rendered body type: text/plain

> But speaking of latency, the real problem is the use of such tight
> synchronicity in the first place. Who says I should design my schemes
> to sound like AMTOR? Why can't I make long transmissions with long

Ok, AMTOR is kind of an extreme example. But you aren't advocating
the complete resynchronisation at the start of every transmissions,
are you? Not even CDMA can afford this.

You cannot tell me that you can do a reasonable modem without
some realtime requirements, and that's inherently more difficult
on an unknown environment such as a general purpose PC operating
system than on a known dedicated environment such as a DSP.

Tom

74. Re: [HFSIG:2280] Re: Serial versus parallel, mainly parallel

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 15 Jul 1997 13:01:50 +0100
To: hfsig@tapr.org
Message-ID: <33CB66AE.3978BC66@dircon.co.uk>
Rendered body type: text/plain

Thomas Sailer wrote:

> > Unfortunately PCs don't last very long when dropped 1.2 metres onto
> > concrete and submerged 1 metre underwater for 4 hours... They don't
> > usually start at -40 degrees and something's usually gone bang
> before
> > +50 degrees. PCs MTBF (Mean time between failures) isn't high enough
>
> > for unattended remote operation, far too many transistors and
> > mechanical whirry things, the silicon runs too hot, and not enough
> screws.
>
> Valid point. That's why we don't use PC's in packet radio node
> controllers, which are often installed in harsh environments, such
> as mountain tops.
>
> I heard stories of people travelling a few hours just to
> hit "F1" so that the PC could reboot :-)
>
> Tom

Well I thought I might have my say.

I have been looking into this subject for a few months now and I feel
that using Intel
technology is becoming the least risk/cheapest method of developing the
hard real
time systems we all love.

Apart from the fact that there are good development tools free on the
Internet i.e
DJGPP PC GNU environment (consisting of 32 bit assembler/linker/compiler
(C C++
Objective C)/archiver/debugger). There are a number of companies selling
386->Pentium
single board computers with flash disks etc that take standard P.C
cards. These can be
incorporated into ruggedised cases etc if you really want. You can even
buy ones with
extended temperature ranges.

Once you make the jump to 32 bit protected mode most of the problems
encountered with
real mode go away and it looks like any other 32 bit processor.

I am also struggling to get RTEMS (a free 32 bit pre-emptive
multitasking Kernel)
which was developed for the U.S army to run on it as well.

Currently I am using a ADSP2181 DSP EZkitLite mounted on a P.C ISA
prototype card
that allows me to talk to the DSP via its IDMA interface. The total cost
was $200.
The board appears to my program as an I/O port and I talk to it/download
software simply
by reading and writing to it. Yes it is a bit of a birds nest, but with
the lid on no-one sees
it. The board has plenty of extra space for relays etc to control the
radio. I used a standard
P.C ISA card with ready made interface logic, a couple of extra TTL
chips and a lot of short
bits of wire, it took about 1 day to wire up. The biggest problem was
lifting a couple of the
legs on the surface mount DSP chip to get it to boot via the IDMA port.

My DSP code was modelled using PC-SIMW/PC-DSPW which saved me a lot of
wasted
effort. The company that produce these items also do a set of C DSP
libraries which you
can use to prototype just about anything in non-realtime.

Eventually the project I am working on will be transferred to the SGC

Smartmodem. This
consists of an i386EX and ADSP2181. This board runs DOS but has extra
Interfacing
for radio control etc. (See their www site). It is designed as an open
architeture with
the idea of encouraging people to develop applications for it. (Sorry
for the commercial
plug!)

That way I get the best of both worlds, my development environment is
cheap and if the
project is sucessful I have ready made commercial hardware for it to run
on.

Other than DSP EVMS, commercial ISA DSP cards are very expensive, 3-5
times the
price of an EVM. One problem with EVMs is they tend to use the first
generation masks
of the various DSP chips which sometimes contain hardware bugs. I
believe this was the
case of the TMS320C50EVM. Anthoney has some experience of this. I guess
it is a good
way for the manufacturers to get rid of unwanted stock.

As far as re-booting is concerned the 386EX has a hardware WDT to allow
auto reeboot
of the hardware under fault conditions (Just like the MARS rover!).

I also believe that we have to look in two directions with our
developments, firstly in
producing something the bulk of amateurs want to use but also towards
the wider public
to show that amateur radio is still a breeding ground of exciting
futuristic ideas, both in the
development but also the explotation of these technologies.

Amateurs complain that the Internet is killing off the hobby, it is not,
it is just changing it.
Without the Internet I couldn't do the types of things I am doing now. I
think the Internet
is the best thing that ever happened to the hobby (since the invention
of SSB).

- Charles G4GUO.

75. Development Platforms (was serial / parallel)

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 15 Jul 1997 13:50:33 +0100
To: hfsig@tapr.org
Message-ID: <33CB7219.395AF5CD@dircon.co.uk>
Rendered body type: text/plain

Bergzen Hakan, KR wrote:

> Hello,
>
> I hope I am excused for jumping into the ongoing title-deviated
> discussion
> like this :-)

Your perfectly welcome Hakan, by the way how did the HF conference go?

>
>
>
> Personally, I would not like to use Linux since all my other programs
> (for
> good or bad) are housed in the MS-Windows environment, and a temporary
>
> switch would make my PC just as specialised as a DSP board.
>

Don't you have more than one P.C ? When you upgrade to your Pentium so
youcan run the latest game you use the old one for Linux!

> with some sort of black box. E.g. the soundblaster i/o will probably
> need
> external circuitry to handle impedance matching and keyline
> signalling.

>

Yes that is the main problem with using standard P.C parts. You can buy
boardsto do all this but they are expensive.

> The two methods have actually more similarities than differencies, ie
> they
> are both based on standardised platforms (EVM starter kits are quite
> standardised really) and the freely available software/hardware gives
> a ham
> a possibility to learn modem functionality and implementation (as
> opposed to
> buying a PTC/Clover/something other modem). In short, they both take
> back
> the development initiative from the ham radio companies to the
> original ham
> operators.
>

Actually I have spoken to three ham manufacturers HAL, SCS and SGC and
allthree were keen for third parties to develop applications using their
hardware.

By applications I mean using the hardware for something else not writing
a
Win95 front end for it.After all, while I am able to write software from
my spare room I don't want to
set up a production line here!

- CUL Charles

76. Re: [HFSIG:2282] Re: Serial versus parallel, mainly parallel

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 15 Jul 1997 08:31:20 -0800
To: hfsig@tapr.org
Message-ID: <199707151528.IAA27747@peak.org>
Rendered body type: text/plain

Tom,

>It's easy. Just grab C compiler and read the appropriate
>"HOWTO". From my experience, the overall gain won't be big
>if you hack anything in asm...
>
>If you want to know how to do audio IO under Linux, read the
>manual at http://www.4front-tech.com/pguide/, if you're interested
>in Windoze, read the "MMSYSTEM" docs from M$, especially
>the WaveIn* and WaveOut* functions. For DOS, search the Web
>for Ethan Brodsky's soundblaster page...

Just a casual observation: you seem to always only release executable code -
seldom do
we see any of your fine handywork's source code. Is there a reason for this?
>From my own personal experiences; you have no idea how much further your
efforts goes
when you share source code - it helps others educate themselves and perhaps
in future
become contributors as well.

73's Johan

77. COHERENT BPSK SOFTWARE

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 15 Jul 1997 08:49:05 -0800
To: HFSIG@tapr.org, DSP@tapr.org
Message-ID: <199707151546.IAA01346@peak.org>
Rendered body type: text/plain

Hi all,

I have written some DSP code that turns the EVM into a front end for the
VE2IQ COHERENT
software. If you are not familiar with the COHERENT program: it does BPSK on HF
or Laser beams at several user-selectable rates and a variety of operating
modes,
several of these contain error-control coding. It is really impressive to

see one print
signals that are down into the noise. You also learn how critical it becomes
to hold frequency
to a few Herz.

With this EVM software, you use the same EVM hardware as before. There is
only one
minor modification needed to use the EVM for this work; the ceramic
resonator has to be
replaced with a real 4.0 MHz crystal. I'll include a short note on doing
this simple improvement.

Watch for the announcement for a software package that I will upload to the
DSP area. This
contains source code, loadable modules, and other bits of information on
where to obtain COHERENT
and where to participate in BPSK activities on HF.

Hope to see some real participation in this fun experimental work.

73,

Johan, KC7WW

78. Re: Dev Platform (was: Serial versus parallel, mainly parallel)

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Tue, 15 Jul 1997 18:31:00 +0200
To: hfsig@tapr.org
Message-ID: <33CBA5C4.4550@ife.ee.ethz.ch>
Rendered body type: text/plain

> Just a casual observation: you seem to always only release executable code -
> seldom do
> we see any of your fine handywork's source code. Is there a reason for this?

My personal attitude has always been to put the Linux code under
the GPL, i.e. freely available (and it is, check out your favorite
Linux Kernel mirror for example), and not to release any source code
for DOS/Windoze.

My reasons are:
1) Linux is the better platform for experimentation, and the code
   is easier to understand because of less interference from
   the OS
2) Windoze is a highly commercial platform, and you don't get the
   sources to the OS anyway

I'm not the only one with this policy: Flori, DL8MBT, for example,
distributes the linux version of his mailbox software in source and
the DOS version binary only

> become contributors as well.
Anyone want to volunteer writing an HF terminal program for X windows?
That'd be needed right now :-)

73s
Tom

79. Re: [HFSIG:2287] Re: Serial versus parallel, mainly parallel

From: Antony Martin <anthony.martin@gecm.com>
Date: Tue, 15 Jul 1997 18:44:51 +0100
To: hfsig@tapr.org
Message-ID: <970715184451.25766@oas01.gmcl.co.uk>
Rendered body type: text/plain

** I've been having trouble posting but it's a DNS problem at my end. The DNS
lookup is overloaded and the mail router can't be bothered to try 6 times **

> Just a casual observation: you seem to always only release executable code -
> seldom do
> we see any of your fine handywork's source code. Is there a reason for this?
> >From my own personal experiences; you have no idea how much further your
> efforts goes
> when you share source code - it helps others educate themselves and perhaps
> in future
> become contributors as well.
>
>

> 73's Johan
>
Yes, why is there so little sourcecode on the www.tapr.org server?

Only faceless object or hex files.

Without sourcecode the essential portability is lost.

It can't be adapted to cheaper codecs or a higher clock rate.

The educational value is lost.

Bugs can't be fixed. Nobody's code is perfect.

Improvements can't be tried.

The Amateur's ability to experiment is decimated.

Plus hex files convey nothing of the personality and
skill of the creator.

It also documents the work that has been done. Documentation
is otherwise very sparse.

It also causes proliferation of standards if others can't find
out what you did and build on it.

The amateurs don't know what they're doing. The licence test is
supposed to stop you being an ameteur if you don't know what you're
doing. You must fully understand the modulation you use, it goes
with the responsibilities of being a licence holder, & with the
rights to experiment so granted by it.

Anthony

ps. Yes Charles, the 'C50 DSKs do have rev.1 chips on them that need
workrounds. We've replaced the chips on 2 DSKs as our debugger won't
support rev.1 devices. One of them runs double-speed at 80MHz now!

If anyone has a problem with this I'll gladly change the chip for you,
I've got a box of 'C5xes under my desk I'm never likely to use as we're
using 3 Volt ones now.

80. Re: [HFSIG:2281] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 15 Jul 1997 11:10:24 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707151810.LAA24583@servo.qualcomm.com>
Rendered body type: text/plain

>(it is much easier for a company to put a higher price on something physical
>that you can actually see, than just software), one always seems to come up

Reminds me of a discussion at a TAPR meeting in Tucson some years back
on this very subject. I pushed for more use of off-the-shelf hardware
with amateur-written software. Somebody in the group said, "But can you
make money on software?" My immediate response: "Ask Bill Gates."

Phil

81. Re: [HFSIG:2284] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 15 Jul 1997 11:18:18 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707151818.LAA24653@servo.qualcomm.com>
Rendered body type: text/plain

>Ok, AMTOR is kind of an extreme example. But you aren't advocating
>the complete resynchronisation at the start of every transmissions,
>are you? Not even CDMA can afford this.

Sure, why not? If you need synchronized clocks, use GPS. Or keep
the synchronicity but just make the transmissions a lot longer.

My point is that AMTOR is a strange protocol design whose ultra-short
"packets" seem to aggravate a lot of problems that are best avoided

by going to longer transmissions.

Phil

82. Re: [HFSIG:2292] Re: Serial versus parallel, mainly parallel

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 15 Jul 1997 20:02:22 +0100
To: hfsig@tapr.org
Message-ID: <33CBC93E.33C39EC@dircon.co.uk>
Rendered body type: text/plain

Phil Karn wrote:

> My point is that AMTOR is a strange protocol design whose ultra-short
> "packets" seem to aggravate a lot of problems that are best avoided
> by going to longer transmissions.
>
> Phil

A long time ago I went to a lecture by Peter Martinez on his new fangledAMTOR system. I seem to recall that the original SITOR
system was
developed by Philips and was implemented with valves. Now that would
be a project and a half.

- Charles

83. Re: [HFSIG:2276] Re: Serial versus parallel, mainly parallel

From: Geoff Bagley <geoff@gcbagley.demon.co.uk>
Date: Tue, 15 Jul 1997 07:17:04 +0100
To: hfsig@tapr.org
Message-ID: <E30xpBAgXxyzEwW+@gcbagley.demon.co.uk>
Rendered body type: text/plain

In message <199707150413.VAA17622@servo.qualcomm.com>, Phil Karn
<karn@qualcomm.com> writes
>>
chances are very good that you're going
>to plug your $20 DSP into the serial port of a Pentium that will
twiddle
>its thumbs for a few million clocks between each bit from the DSP!
>
>
>
>But for small-scale ham experimentation and self-education......
>
>Phil
>
>
>

I like my little TMS320C50.
It lets me get further away from Windows :-)

--
Geoff Bagley
G3FHL.

84. Re: [HFSIG:2288] COHERENT BPSK SOFTWARE

From: sscalsk@mail.ameritel.net (Stan Scalsky)
Date: Tue, 15 Jul 1997 20:47:00 +0600
To: hfsig@tapr.org
Message-ID: <2.2.16.19970715144700.2b0ffdfa@mail.ameritel.net>
Rendered body type: text/plain

About this BPSK signal and comments made by Johan
"It is really impressive to see one print signals that are down
into the noise. You also learn how critical it becomes to hold
frequency to a few Herz."

For a signal that is "down in the noise" are their any techniques
for finding such a signal. What if I miss the ID say maybe given
by CW? Can these type of signals be detected readily enough so
I can find them quickly.

It used to be easy finding signals by ear but with Spread Spectrum
and signals such as this its getting so that everything sounds like
noise or is buried in the noise.

Look forward to the software release.

-= stan


```
--------------------------------------------------------------------
Worldwide UTE News Club                   WUN list admin
code3list and code30users                 owner/list admin
Keeper of the Digital Signals FAQ         latest: V4.0

Primary email: sscalsk@mail.ameritel.net  PGP key:(v2.6.2) CF041381
Backup  email: sscalsk@erols.com          none
```


85. Re: [HFSIG:2295] Re: COHERENT BPSK SOFTWARE

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 15 Jul 1997 23:33:02 -0800
To: hfsig@tapr.org
Message-ID: <199707160629.XAA06348@peak.org>
Rendered body type: text/plain

Stan,

>For a signal that is "down in the noise" are their any techniques
>for finding such a signal. What if I miss the ID say maybe given
>by CW? Can these type of signals be detected readily enough so
>I can find them quickly.

These signals are quite recognizable after using it for a while. Even the baudrate and the error-correcting scheme becomes familiar with a bit of practise. One usually have a good clue which option to turn on the see if it prints.

Earlier this evening, I worked VE2IQ using 10W on 40M with my vertical antenna. I'm located on the West Coast and he is in Quebec, Canada - not bad. I could just barely hear his signal, actually I just finished up a QSO with a station in Santa Barbara and was doing something else in the room, when I looked on the screen there was someone calling me - I could not hear the signal! When I paid attention and cranked the audio up, I could hear him come up a little above the static crashes, SSB and CW QRM. We selected the strongest ECC, and although the throughput slowed down to a crawl, we had a nice QSO. I was thrilled!

>
>Look forward to the software release.

I'm cleaning up the code a little and should post the package by the weekend.

--Johan

86. Re: [HFSIG:2282] Re: Serial versus parallel, mainly parallel

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Wed, 16 Jul 1997 13:59:11 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199707161059.NAA19779@boojum.hut.fi>
Rendered body type: text/plain

> It's easy. Just grab C compiler and read the appropriate
> "HOWTO". From my experience, the overall gain won't be big
> if you hack anything in asm...
>
> If you want to know how to do audio IO under Linux, read the
> manual at http://www.4front-tech.com/pguide/, if you're interested
> in Windoze, read the "MMSYSTEM" docs from M$, especially
> the WaveIn* and WaveOut* functions. For DOS, search the Web
> for Ethan Brodsky's soundblaster page...

Thanks. I think this approach becomes really popular after there is a "software box" (hopefully .rpm) available to all Linux users. I can just imagine istalling the package and getting nice RTTY terminal with a simple and cheap hardware interface to the radio. Right now I am just trying to learn enough Tcl/Tk to control my radio. It will take a while before I could imagine doing the real modem applications in PC.

Jaakko/OH7BF

87. RE: [HFSIG:2296] Re: COHERENT BPSK SOFTWARE

From: jalocha@axdele.cern.ch
Date: Wed, 16 Jul 1997 14:40:44 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199707161240.OAA19136@dxmint.cern.ch>
Rendered body type: text/plain

Johan, could you please say a word on the kind of ECC schemes used
the COHERENT software and on the protocol between the PC and the DSP,
what kind of information is being passed there ?

Pawel

88. Re: PC modem implementation (was: Serial versus parallel, mainly parallel)

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Wed, 16 Jul 1997 15:32:51 +0200
To: hfsig@tapr.org
Message-ID: <33CCCD83.4A6D@ife.ee.ethz.ch>
Rendered body type: text/plain

Jaakko Koivuniemi wrote:
> Thanks. I think this approach becomes really popular after there is
> a "software box" (hopefully .rpm) available to all Linux users. I can

It won't be as easy as that. Some popular HF modes such as
Amtor and Pactor require relatively accurate clocks (specifications
say within +-20ppm, although most implementations will also work
with +-100pm). None of the clock sources inside a PC is usually
that accurate, so there is a need to calibrate clocks. And I have
yet to find an easy method to calibrate those clocks.
The current method is to use the german longwave DCF77 transmitter,
which somehow reduces the geographic region such a solution is
useful :)

Another idea was to use the line sync of a TV signal

If someone has another idea, please contact me :)

Tom

89. Re: [HFSIG:2298] Re: COHERENT BPSK SOFTWARE

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 16 Jul 1997 08:09:47 -0800
To: hfsig@tapr.org
Message-ID: <199707161506.IAA28014@peak.org>
Rendered body type: text/plain

Pawel,

>Johan, could you please say a word on the kind of ECC schemes used
>the COHERENT software and on the protocol between the PC and the DSP,
>what kind of information is being passed there ?
>
>Pawel
>
>

A bit of background.....

All data frames are synchronous; in the uncoded ASCII case, 10-bit
characters (1 start, 2 stop, 7 data) are sent - when the buffer is empty, it
sends nulls to keep the timing straight. The software does its
syncronisation inially (epoch sync) by hunting for 1-1-0 (last stop pattern,
next start bit). Once synced, it never looks at the start/stop bit again as
all timing is then derived internally (remember what I said about the EVM's
ceramic resonator not being good enough ... with a crystal replacement, it
will hold sync indefinately, even when the signal dissapears for a minute).

There are two ECC schemes called ET 1 and ET 2; These are simply a mapping
of the ASCII symbol to either a 16-bit, or a 27-bit binary code
respectively. If you have worked or read about Leech Lattices this would be
familiar - the codes are found by maximizing the "distance" between code
elements - similar problem as the famous "ball packing" problem. Although
the implementation is highly experimental, it is possible to do fairly
sophisticated error-detection and correction by deriving syndromes by use of
sub lattices. This however, is not quite a simple ... better read a bit
about those things ... I'm no expert anyway!

Bill, VE2IQ has done program rather elegantly; working his way up from first
principles to get a good handle on weak signal demodulation,

synchronisation, and now is playing with ECC. He is toying with the ieda of
incorporating diversity measures in the near future and probably will
eventaully progress on to other things like convolutional code, I am sure.

The COHERENT program derives all its internal timing from an external
interface that provides differentally-encoded 8-bit data at 7200 SPS. This
data is passed to COHERENT via a COM port at 115.2KB. It is critical that
the 7200 SPS rate be very (I mean very) stable and accurate.

Anyway, you should download the software package and notes from Bill's web
site and read first hand how it all works: http://ietc.ca/home/bill/bbs.htm

Hope this helps.

--Johan

90. Re: [HFSIG:2298] Re: COHERENT BPSK SOFTWARE

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 16 Jul 1997 08:16:15 -0800
To: hfsig@tapr.org
Message-ID: <199707161513.IAA28992@peak.org>
Rendered body type: text/plain

Pawel,

>Johan, could you please say a word on the kind of ECC schemes used
>the COHERENT software and on the protocol between the PC and the DSP,
>what kind of information is being passed there ?
>
>Pawel
>
>

>Anyway, you should download the software package and notes from Bill's web
site and read first hand how it all >works: http://ietc.ca/home/bill/bbs.htm

Correction: http://www.ietc.ca/home/bill/bbs.htm

--Johan

91. RE: [HFSIG:2300] Re: COHERENT BPSK SOFTWARE

From: jalocha@axdele.cern.ch
Date: Wed, 16 Jul 1997 18:12:19 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199707161612.SAA26060@dxmint.cern.ch>
Rendered body type: text/plain

Thanks Johan, I need one detail to be explained:

>The COHERENT program derives all its internal timing from an external
>interface that provides differentally-encoded 8-bit data at 7200 SPS. This
>data is passed to COHERENT via a COM port at 115.2KB. It is critical that
>the 7200 SPS rate be very (I mean very) stable and accurate.

That is the character rate on the serial port must be stable
or the bit rate ?
The DSP processing is synchronized to the CODEC sampling (freq. = F)
(the DSP only needs to be fast enough) and if the DSP outputs
a processed data sample every Nth audio sample then my naive expectation
is that the serial port must only be faster than F/N and it is the CODEC's
sampling which we should keep stable.
Now you say, you care for the DSP clock and not for the CODEC clock, so please
explain this detail to me :-)

Pawel

92. Re: [HFSIG:2302] Re: COHERENT BPSK SOFTWARE

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 16 Jul 1997 12:22:20 -0800
To: hfsig@tapr.org
Message-ID: <199707161919.MAA16173@peak.org>
Rendered body type: text/plain

>Thanks Johan, I need one detail to be explained:
>
>>The COHERENT program derives all its internal timing from an external
>>interface that provides differentally-encoded 8-bit data at 7200 SPS. This

>>data is passed to COHERENT via a COM port at 115.2KB. It is critical that
>>the 7200 SPS rate be very (I mean very) stable and accurate.
>
>That is the character rate on the serial port must be stable
>or the bit rate ?

COHERENT needs a precise 7200 SPS clock ... it gets that from receiving
character interrupts from
the COM port, i.e., 7200 interrupts per second. Further, the COM port must
run at 115.2KB. These
parameters are fixed - you must not change those.

So, whatever drives your character sending rate, CODEC or Timer, is
immaterial, as long as it meets
above criterium. Simple enough?

--Johan

93. unsubscribe hfsig

From: "Norman R. Johnson" <normj@interaccess.com>
Date: Wed, 16 Jul 1997 19:06:16 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BC921B.56D4F320@hp>
Rendered body type: text/plain

[Empty message body]

94. subscribe hfsig

From: Kaprocki <fkaprocki@iag.net>
Date: Wed, 16 Jul 1997 20:51:29 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970717005129.00665b14@iag.net>
Rendered body type: text/plain

[Empty message body]

95. Re: [HFSIG:2299] Re: PC modem implementation (was: Serial versus parallel, mainly parallel)

From: Lyle Johnson <lylej@azstarnet.com>
Date: Wed, 16 Jul 1997 19:48:33 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19970716194802.006f9a5c@pop.azstarnet.com>
Rendered body type: text/plain

>If someone has another idea, please contact me :)

Tha TAPR TAC project uses GPS to provide a clock. The follow-on (there we
go with dedicated hardware again, why not just use a virtual clock :-)
project will be the TOC< which is a disciplined 10 MHz oscillator sync'ed
to GPS to provide a reference accurate to about 1 part 10^9.

Cheers,

Lyle

96. Re: [HFSIG:2283] Re: Serial versus parallel, mainly parallel

From: Lyle Johnson <lylej@azstarnet.com>
Date: Wed, 16 Jul 1997 19:48:30 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19970716193412.0076bf14@pop.azstarnet.com>
Rendered body type: text/plain

>> But for small-scale ham experimentation and self-education, I could
>> never for the life of me understand the ham (TAPR?) inclination to
>> spend considerable resources designing, building and replicating
>
>That's something I didn't understand too... There are plenty of
>cheap platforms around to experiment with, so why build
>yet another one?

The answers are easy (and not complete):

1) it's fun for some of us to design and implmenet hardware, just like it
is fun for some of us to design and implment software;

2) if I make a silly mistake and zap my $149 EVM, I am only out a $149 EVM
and can continue to play around on the Web, simulate systems, etc., on my
still-functional PC;

3) my 200 Mhz Pentium Pro isn't very portable, but my ancient Toshiba T1000 (or my Tandy 100, or NEC 8201A or Atari Portfolio or...) is, and draws little power and I can take it camping and run it from solar panels, and the EVM draws little enough power that it can, too. This cheap, low-risk, low-power combination can provide lots of enjoyment and if the bears eat it, I'm not out much. Not so my $3,800 Gateway Solo P166 MMX laptop...;

4) if I want to learn more, the source code for the EVM applications, at least, is freely available (as is Phil's code). It isn't obscured by lots of calls to Windoze APIs, or filesystem magic, or any other complication. It is clean, focussed and understandable. Many PC-based (and, sad to say, ADI and TI DSP) ham radio applications are only object-code files that can be downloaded and used, but not learned from;

5) Like the PC, good tools for the EVM are free (unlike TI and ADI, again, sadly);

6) simple hardware is easy to use and understand at the lowest levels. PCs are complex beasts full of undocumented ASICs that have lots of undocumented registers. The only way most people can begin to use them is to write high-level applications using canned I/O packages and hope that they aren't stepping on some interrupt vector (and try to find an available interrupt to use a card in a slot to do something new). The whole computer acquires a sort-of mystique that only the 'specially anointed can delve into -- the whole machine is just too complex for most folks to figure out and do useful things at a low level.

And for many folks, that PC represents a *significant* investment. A grand or two isn't spent lightly, nor risked easily.

Do you have a schematic for your PC (not your orignal 5-slot IBM box gathering dust in the corner, it had a useful tech reference manual for $30 and was made out of parts, not ASICs)? Is it useful?

>> quite willing to use it for the application in question, at least on
>> an experimental basis, before blowing money on specialized hardware.
>
>IMNSHO the average ham doesn't understand much of it, so he's thinking
>that a box that costed 1000$ must perform better than free
>software... the same argument basically as why companies use
>NT instead of Linux :-)

Well, we use both. Linux for software development, because it is excellent at this. We use NT/Win95 for hardware development because we don't want to write Xilinx FPGA mappers, we want to map Xilinx FPGAs (or draw schematics, or layout PCBs, or...), and the EDA tools that are affordable for small and medium sized companies happen to run under Win95 and NT. If they ran under Linux we'd switch in a heartbeat.

Oh, and we use Slowaris on the SUN Ultras because that's what the company runs its busniess software one.

I think you'll find that many companies use what works to solve the problem...it isn't an O/S religious issue, it's a cost-effective-way-to-solve-the-problem issue...

Just my two pfennigs,

Lyle

97. Re: [HFSIG:2259] BPSK & Parallel modem tests status:

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 16 Jul 1997 21:53:11 -0800
To: hfsig@tapr.org
Message-ID: <199707170449.VAA01805@peak.org>
Rendered body type: text/plain

Dear Eduardo,

Sorry for the delay in responding.

I have not done much PSK beacon work after the last try to work Europe. From time to time I do listen for the CW beacons on 14 and 21 MHz, but so far nothing much to report on.

A friend of mine, however, tells me he works into Europe on a regular basis, but he has an excellent antenna. That tells me that one can do it if you are prepared to invest a little in preparing equipment. Neither I or,

apparently, anyone from the European group is properly equipped it seems.

Recently, I have established contact with another group of PSK
experimentors, and now have a program for the EVM that works nicely with
their system. I have had contacts around the USA and my experience of how
this mode works has been getting better every day. It really is quite
successful, but it requires that one use a radio with good frequency
stability and resolution to 1 Hz (it will work to +/- 5 Hz). I invested in a
stabilized crystal and 500 Hz crystal filter for my radio, which seems to
have helped a great deal.

I think we need to wait for better conditions, or an oppotune time when the
bands are usable. I would very much like to see a contact on 21 or 28 Mhz.

That's my news, take care and 73

--Johan

98. Re: [HFSIG:2306] Re: PC modem implementation (was: Serial

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Thu, 17 Jul 1997 10:15:43 +0200
To: hfsig@tapr.org
Message-ID: <33CDD4AF.744D@ife.ee.ethz.ch>
Rendered body type: text/plain

Lyle Johnson wrote:
>
> >If someone has another idea, please contact me :)
>
> Tha TAPR TAC project uses GPS to provide a clock. The follow-on (there we
> go with dedicated hardware again, why not just use a virtual clock :-)
> project will be the TOC< which is a disciplined 10 MHz oscillator sync'ed
> to GPS to provide a reference accurate to about 1 part 10^9.

I expected someone would come up with GPS, so here's why I don't
like GPS for the application:

1) Neither 10MHz nor 1Hz do fit into the passband of a PC soundcard
   (that's important!)
2) GPS is expensive; I already own a TV receiver and an HF transceiver
   (obvious if one wants to do pactor :)), but not a GPS receiver
3) I don't want to get dependent on the US govt :)

73s
Tom

99. Re: Serial versus parallel, mainly parallel

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Thu, 17 Jul 1997 10:46:47 +0200
To: hfsig@tapr.org
Message-ID: <33CDDBF7.4E50@ife.ee.ethz.ch>
Rendered body type: text/plain

> 2) if I make a silly mistake and zap my $149 EVM, I am only out a $149 EVM
> and can continue to play around on the Web, simulate systems, etc., on my
> still-functional PC;

That was kind of a misunderstanding, my remark was targeted
at the DSP93 effort, and the "cheap platforms" included the
Mot EVM :)

> 4) if I want to learn more, the source code for the EVM applications, at
> least, is freely available (as is Phil's code). It isn't obscured by lots
> of calls to Windoze APIs, or filesystem magic, or any other complication.
> It is clean, focussed and understandable. Many PC-based (and, sad to say,

Have you taken a look at my Linux hamradio modems? They aren't
obstructed by windoze API calls :) Do you think they're harder
to understand than EVM software? If so, what should I do to
make it more understandable?

> 5) Like the PC, good tools for the EVM are free (unlike TI and ADI, again,
> sadly);

My definition of a good C compiler is somehow different. I use
the EVM at work. And the inefficiency of the C compiler drove me
into writing my "OS kernel" in Asm.

> 6) simple hardware is easy to use and understand at the lowest levels. PCs
> are complex beasts full of undocumented ASICs that have lots of

That used to be so a few years ago. But now the docs aren't worse
than typical EVM docs and available on the web... Now explain me
why the CS4215 on the EVM is better documented in your opinion
than the CS4236 on a cheap PC soundcard...

> Do you have a schematic for your PC (not your orignal 5-slot IBM box

Yes. I used to have Triton docs.

> gathering dust in the corner, it had a useful tech reference manual for $30
> and was made out of parts, not ASICs)? Is it useful?

It was not necessary. I only needed the soundcard docs (CS4236) and
the DMA controller docs (8237, grad an Intel databook, no matter
how old it is :)) to do useful things (unter Linux, that is).
If you don't do timing critical stuff (i.e. something that needs
fast rx/tx switching times), its much easier than that, you
don't need any hw docs at all, OSS's sound programming docs
are enough...

> Well, we use both. Linux for software development, because it is excellent
> at this. We use NT/Win95 for hardware development because we don't want to
> write Xilinx FPGA mappers, we want to map Xilinx FPGAs (or draw schematics,
> or layout PCBs, or...), and the EDA tools that are affordable for small and
> medium sized companies happen to run under Win95 and NT. If they ran under
> Linux we'd switch in a heartbeat.

You might try LOLA and the associated Trianus/Hades system
from N.Wirth from the CS dept of the ETH :) It runs on many
platforms and is free; but granted, it doesn't support a big
variety in different FGPAs

> Just my two pfennigs,
If you addressed that to me - sorry, wrong currency

73s
Tom

100. Re: [HFSIG:2309] Re: PC modem implementation (was: Serial

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Thu, 17 Jul 1997 04:14:03 -0500
To: hfsig@tapr.org
Message-ID: <v03020905aff3927b8258@[208.134.134.40]>
Rendered body type: text/plain

>Lyle Johnson wrote:
>>
>> >If someone has another idea, please contact me :)
>>
>> Tha TAPR TAC project uses GPS to provide a clock. The follow-on (there we
>> go with dedicated hardware again, why not just use a virtual clock :-)
>> project will be the TOC< which is a disciplined 10 MHz oscillator sync'ed
>> to GPS to provide a reference accurate to about 1 part 10^9.
>
>I expected someone would come up with GPS, so here's why I don't
>like GPS for the application:
>
>1) Neither 10MHz nor 1Hz do fit into the passband of a PC soundcard
>   (that's important!)

>2) GPS is expensive; I already own a TV receiver and an HF transceiver
>   (obvious if one wants to do pactor :)), but not a GPS receiver

Hum -- $99 isn't expensive for a GPS ? See http://www.tapr.org/gps offer
ends August 30th!

>3) I don't want to get dependent on the US govt :)

Well, GPS will be here for a long, long time based on the applications
being developed by the DoT and other agencies.

Cheers - Greg

Greg Jones, WD5IVD
Austin, Texas

wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd
-----

101. Re: [HFSIG:2309] GPS/GLONASS receivers

From: "NIKOS J. FARSARIS" <njf@lissos.tsinet.gr>
Date: Thu, 17 Jul 1997 11:46:55 -0200
To: hfsig@tapr.org
Message-ID: <33CE224F.5601@lissos.tsinet.gr>
Rendered body type: text/plain

Thomas Sailer wrote:

> 2) GPS is expensive; I already own a TV receiver and an HF transceiver
>   (obvious if one wants to do pactor :)), but not a GPS receiver
> 3) I don't want to get dependent on the US govt :)
>
> 73s
> Tom

2) No!! you can get a GPS receiver not bigger than a cellural
phone for about $ 500

3) You can switch to the Glonass then or like me use both systems.
I think that this is a better way to overcome dependence of ANY
government.

In smaller areas you can also use VOR signals to get your location
with a CEP of 50-200 meters, or known position radiolinks !!!!
--
NIKOS J. FARSARIS,   Dipl. Elec. Eng., Ph.D. Candidate A.U.TH.
RADAR / RF Research,TELECOMMUNICATION SYSTEMS INSTITUTE OF CRETE
37 Iroon Polytechneiou Ave,73133 CHANIA CRETE GREECE
Tel: 30-821-28457/28423 Fax: 28459 E-mail: njf@tsinet.gr, njf@ee.auth.gr

102. Re: [HFSIG:2299] Re: PC modem implementation (was: Serial versus parallel, mainly parallel)

From: zs6awk@global.co.za (Danie Brynard)
Date: Thu, 17 Jul 1997 13:36:31 +0200
To: hfsig@tapr.org
Message-ID: <199707171135.NAA25501@mail.global.co.za>
Rendered body type: text/plain

And WWV ?

>Jaakko Koivuniemi wrote:
>> Thanks. I think this approach becomes really popular after there is
>> a "software box" (hopefully .rpm) available to all Linux users. I can
>
>It won't be as easy as that. Some popular HF modes such as
>Amtor and Pactor require relatively accurate clocks (specifications
>say within +-20ppm, although most implementations will also work
>with +-100pm). None of the clock sources inside a PC is usually
>that accurate, so there is a need to calibrate clocks. And I have
>yet to find an easy method to calibrate those clocks.
>The current method is to use the german longwave DCF77 transmitter,
>which somehow reduces the geographic region such a solution is
>useful :)
>
>Another idea was to use the line sync of a TV signal
>
>If someone has another idea, please contact me :)
>
>Tom
>
>

++
Danie Brynard   Email: zs6awk@global.co.za
PO Box 15133,Lyttelton,0140,South Africa
GRID LOC: KG44CE,   Tel/Fax: (2712) 6640330
++

103. 64 tone, 10 baud, BPSK modem

From: jalocha@axdele.cern.ch
Date: Thu, 17 Jul 1997 14:45:52 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199707171245.OAA11709@dxmint.cern.ch>

Rendered body type: text/plain

Hi all,

I have just made a working code for the 64-tone, 10 baud BPSK modem.

It sends 64 tones, spaced by 15 Hz, allocated in the 500-1500 Hz audio range.
Each tone is DBPSK modulated at 10 baud thus we get 64-bit data blocks
and 10 blocks per second. User data is organized as 7-bit ASCII character
stream. Each character is encoded with Walsh functions onto a 64-bit codeword.
The codewords are then interleaved which spreads each character over about
3 seconds. Overall user data rate is thus 10 7-bit ASCII characters per second.

The modem audio input/output design is based on 8000 Hz sampling
and an anti-alias filter which passes no more than 1600 Hz around the
1000 Hz point. The data is down sampled by 5 (8000/5 = 1600 Hz)
and then follows a 512 point FFT with sliding window.

The demodulator is a pure phase approach: after the FFT I turn I/Q vectors
into phases and all following analysis is done on these phases.
I did so due to memory constraints of the EVM56K (I need to keep
quite some history of the FFT slices) and because it made the code
a bit simpler (you only do phase addition, subtraction, averaging, etc.).
The demodulator computes soft decisions and the decoder is done with the
fast (soft) Hadamard transform.

The receiver can lock within about +/- 50 Hz and the lock range can be made
larger in principle. There are no dedicated tuning/sync. tones - the receiver
is able to lock within 5-10 seconds onto an ongoing transmition.
After the lock the receiver tracks the frequency drift (up to some 0.05Hz/sec
I think) and the timing drift too.

The user interface is RTTY-like: Ctrl-T turns the transmitter on,
Ctrl-R turns it off. Any character sent to the DSP is transmitted
(if the transmitter is off, the data is buffered),
whatever the receiver captures is sent to the terminal
(after a signal is acquired).

Who will try it first on the air ? :-)

Pawel

104. Re: [HFSIG:2309] Re: PC modem implementation (was: Serial

From: Lyle Johnson <lylej@azstarnet.com>
Date: Thu, 17 Jul 1997 07:38:33 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19970717072427.0074daf4@pop.azstarnet.com>
Rendered body type: text/plain

At 03:20 AM 7/17/97 -0500, you wrote:
>Lyle Johnson wrote:
>>
>> >If someone has another idea, please contact me :)
>>
>> Tha TAPR TAC project uses GPS to provide a clock. The follow-on (there we
>> go with dedicated hardware again, why not just use a virtual clock :-)
>> project will be the TOC< which is a disciplined 10 MHz oscillator sync'ed
>> to GPS to provide a reference accurate to about 1 part 10^9.
>
>I expected someone would come up with GPS, so here's why I don't
>like GPS for the application:
>
>1) Neither 10MHz nor 1Hz do fit into the passband of a PC soundcard
>   (that's important!)
>2) GPS is expensive; I already own a TV receiver and an HF transceiver
>   (obvious if one wants to do pactor :)), but not a GPS receiver
>3) I don't want to get dependent on the US govt :)
>
>73s
Tom,

But you didn't ask for "a better idea that doesn't include GPS" just for a
"better idea" :-)

1) Add a divider from the 10 MHz clock, to 1 kHz or whatever you like. It
will still be good to 1 part in 10^9. Or, use a PLL multiplier and sync it
off the 1 Hz output from the GPS receiver and save some expense, but you
might degrade your accuracy to 1 part in 10^6 or 10^7. Good enough for
sound card sync, no?

2) TAPR has a $99 GPS receiver special buy (includes antenna). Lots
cheaper than a TV (and more interesting to watch, no commercials!) and uses
MUCH less power :-) Add the PLL sync circuit mentioned above for $10 or so
and you've got a dirt cheap, accurate audio source for your sound card.
Pick your tone, your reference for the PLL/SYNC is 1Hz, so any audio tone
that makes it easy for the algorithm you wish to implement is yours for $10
or so.

3) You can spend a lot more francs and get a GLONASS receiver if being
anti-US is important to you. The German time signal you referred is
probably coming from someone's government since there isn't much money to
be made commercially in sending time signals (just in receiving and
manipulating them :-)

Actually, the advantage of a GPS (or GLONASS) scheme is that it *IS*
global, not geographically restricted (except you can't use it underground
and you need an outdoor antenna -- but then so do you in general for HF
comms), it'll probably be standard equipment on your next new car, the
signals are FREE,...

Since I'm a US taxpayer, it's a chance to get something back from the
gov't! And, I'm happy to share the signals I invested in with you. Think
of it as coming from a fellow Ham, not the governement!

Have fun!

Lyle

105. Re: [HFSIG:2310] Re: Serial versus parallel, mainly parallel

From: Lyle Johnson <lylej@azstarnet.com>
Date: Thu, 17 Jul 1997 07:38:36 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19970717073534.0076935c@pop.azstarnet.com>
Rendered body type: text/plain

Tom,

My remakrs were addressed to the general audience on the SIG, not to any
one in particular. Ths issue rasied was two people not understanding why
TAPR and others wanted to build hardware.

The EVM success appears to be in large part due to the pioneering work and
success of the ALef Null group, and the fact that the EVM was so close in
design to the DSP Card 4, that the DSP Card apps got ported to the EVM.
Its roots were in a non-commercial, amateur endeavor.

My ADI and TI remarks were based on most of what I have seen on the web,
not the efforts of any one particular person.

My PC remarks were directed at the PC, not the sound card codec. The PC
itself is complex and its internals are buried from the eyes of most people.

My remark about "good tools" for the EVM were recognizing the
Motorola-supplied tools available on their website versus the $$$ ADI and
TI tools. SUre, there are "Shareware" assemblers for these, but then one
gets wrapped up in incomatpible syntaax issues (like trying to write an app
that works with both MicroSoft and Borland C). Yes, the Gnu tools are
probably better anyway :-)

I apologize if you thought anything I wrote was directed at you, I was just
adding comments to a gereral discussion of why some folks like to build
hardware.

Cheers,

Lyle

106. Re: [HFSIG:2311] Re: PC modem implementation (was: Serial

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Thu, 17 Jul 1997 17:04:13 +0200
To: hfsig@tapr.org
Message-ID: <33CE346D.1955@ife.ee.ethz.ch>
Rendered body type: text/plain

Greg Jones, WD5IVD wrote:

> Hum -- $99 isn't expensive for a GPS ? See http://www.tapr.org/gps offer
> ends August 30th!

Thanks for answering the less important points :)

I still fail to see how buying a GPS receiver for more than 100
bucks and soldering some circuit to divide the 10MHz clock
(or whatever leaves the GPS rx) into something a soundcard can handle
should be simpler and cheaper than connecting the soundcard
to a TV receiver...

73s
Tom

107. Re: [HFSIG:2314] 64 tone, 10 baud, BPSK modem

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 18 Jul 1997 08:34:09 -0800
To: hfsig@tapr.org
Message-ID: <199707171534.IAA02129@peak.org>
Rendered body type: text/plain

Thanks Pawel - Great job!

This sounds interesting, I'll give it a try on the 40M BPSK net.

--Johan

>Hi all,
>
>I have just made a working code for the 64-tone, 10 baud BPSK modem.
>
>It sends 64 tones, spaced by 15 Hz, allocated in the 500-1500 Hz audio range.
>Each tone is DBPSK modulated at 10 baud thus we get 64-bit data blocks
>and 10 blocks per second. User data is organized as 7-bit ASCII character
>stream. Each character is encoded with Walsh functions onto a 64-bit codeword.
>The codewords are then interleaved which spreads each character over about
>3 seconds. Overall user data rate is thus 10 7-bit ASCII characters per second.
>

108. Re: [HFSIG:2317] Re: PC modem implementation (was: Serial

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Thu, 17 Jul 1997 11:48:04 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.96.970717114447.21857i-100000@arctic>
Rendered body type: text/plain

On Thu, 17 Jul 1997, Thomas Sailer wrote:

> I still fail to see how buying a GPS receiver for more than 100
> bucks and soldering some circuit to divide the 10MHz clock
> (or whatever leaves the GPS rx) into something a soundcard can handle
> should be simpler and cheaper than connecting the soundcard
> to a TV receiver...

1) There is a 1 PPS output that you connect to one of the unused serial
port handshaking lines, such as CTS. Solder 1 wire.

2) You get worldwide synchronism. Not just the 50 mile radius around your
TV station.

109. Re: [HFSIG:2318] Re: 64 tone, 10 baud, BPSK modem

From: "Edson W.R. Pereira" <ewp@usa.healthnet.org>
Date: Thu, 17 Jul 1997 13:14:50 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.95.970717131405.1779B-100000@satellife.healthnet.org>
Rendered body type: text/plain

On Thu, 17 Jul 1997, Johan Forrer wrote:

> Thanks Pawel - Great job!
>
> This sounds interesting, I'll give it a try on the 40M BPSK net.
>

Which frequencies are you guys using for BPSK experiments?

-- Edson

110. Oops! Wrong words!

From: "Lyle V. Johnson" <johnson@mmsi.com>
Date: Thu, 17 Jul 1997 12:52:19 -0700
To: hfsig@tapr.org
Cc: Lyle Johnson <lylej@azstarnet.com>
Message-ID: <33CE77F3.56A2@mmsi.com>
Rendered body type: text/plain

Just realized I said "anti-US" in a previous post when what I meant was
"non US government".

I apologize for any misunderstandings this may have caused. It was not
my intent... :-(

Lyle

111. Re: [HFSIG:2320] Re: 64 tone, 10 baud, BPSK modem

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 18 Jul 1997 13:03:08 -0800
To: hfsig@tapr.org
Message-ID: <199707172003.NAA12616@peak.org>
Rendered body type: text/plain

Edson,

>
>
>On Thu, 17 Jul 1997, Johan Forrer wrote:
>
>> Thanks Pawel - Great job!
>>
>> This sounds interesting, I'll give it a try on the 40M BPSK net.
>>
>
>Which frequencies are you guys using for BPSK experiments?

7.08100 or 3.59100 MHz (carrier). Lately, most evenings around 7pm PST.
Come join in please.

--Johan

112. Re: [HFSIG:2306] Re: PC modem implementation (was: Serial

From: Antony Martin <anthony.martin@gecm.com>
Date: Thu, 17 Jul 1997 21:03:29 +0100
To: hfsig@tapr.org
Message-ID: <970717210329.12361@oas01.gmcl.co.uk>
Rendered body type: text/plain

> >If someone has another idea, please contact me :)
>
> Tha TAPR TAC project uses GPS to provide a clock. The follow-on (there we
> go with dedicated hardware again, why not just use a virtual clock :-)
> project will be the TOC< which is a disciplined 10 MHz oscillator sync'ed
> to GPS to provide a reference accurate to about 1 part 10^9.
>
> Cheers,
>
> Lyle
>
What?! Geez! I don't believe you wrote that. This kind of accuracy is quite
unnecessary and pointless and and and, quite honestly words fail me.

An ordinary crystal oscillator is quite adequate for any HF modem.
The effects of errors of 100ppm are easily corrected on any modem
waveform. Crystals designed for operation over 10 to 40 degrees C
can easily do +-20ppm.

100ppm error on say a 2kHz input frequency is 0.2Hz. Hardly a problem
as any properly implemented modem will track and correct static errors
up to +-100Hz. If you have say 20ms symbols, after 1000 symbols
(20 seconds) you will be 2ms out if you didn't do any tracking at all.

As on a HF modem you will be doing frequency tracking and you will
be tracking transmitter symbol timing, there's absolutely no point in
the expense of GPS and ovenized oscillators for a modem.

Ovenised oscillators are pretty standard though in synthesized
transceivers which are accurate to 0.1ppm. This gives frequency offsets
of worst-case 3Hz at 30MHz. You may well have a TCXO which should

be good for 1ppm in which case the error is worst-case 30Hz at 30MHz.

How many people have manually tuned rigs these days? Without a
digital frequency display?

The days of the old RTTY tuning indicator are extinct!!! Dead, gone,
buried, RIP, redundant, obsolete, history, exit stage left, alas poor
yorick I knew thee well, life extinct, I tell you!!!

If you're having problems with frequency offset, then go to your DSP
software supplier and say "this software doesn't work, please fix
it", or words to that effect. There's absolutely NO excuse.

Is it a case of 'can't be bothered' or 'don't know how'? If someone
says this modem or that modem has a blinding 0.01b/s data rate and
works down to a -137dBs SNR but won't cope with 100Hz frequency offset
then laugh at them very much. The sooner Hams stop putting up with it
the better. DSPs can still do a very good job of finding and tracking
signals "below the noise", and do it perfectly well without human
intervention.

I'd agree that ceramic resonators are of little practical use whatever
for modems as the're always several percent off. It just makes life
more difficult all for want of a 30 pence crystal. Most audiophiles
would complain if their CD player was that far off. An EVM card with
a ceramic resonator wouldn't be suitable for, well, almost anything
really (I was going to say Ham use). Even the Texas DSKs have a
crystal, the PC internal EVMs are normally higher spec.

I'm seriously considering doing a modem for release to Hams.
The ones I do for Marconi, of course, can't be given away.
I didn't think I could do that much better than the established
ones but I'm starting to have a whole bunch of second thoughts...
they don't look so good after all.

Anthony.

113. unsubscribe hfsig

From: Wolf Baron <wbaron.hondutel.hn@hondutel.hn>
Date: Thu, 17 Jul 1997 15:20:52
To: hfsig@tapr.org
Message-ID: <3.0.2.16.19970717152052.1dcf72d2@hondutel.hn>
Rendered body type: text/plain

please don`t send more mail! 73 - Wolf

114. Re: [HFSIG:2299] Re: PC modem implementation (was: Serial versus parallel, mainly parallel)

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 17 Jul 1997 15:07:24 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707172207.PAA18613@servo.qualcomm.com>
Rendered body type: text/plain

Although I agree with Antony that highly accurate clocks just aren't
needed for HF modems, for completeness I should point out that there's
still another way to get your PC's clock synchronized, one that
doesn't require any hardware at all: NTP, the Internet Network Time
Protocol.

With NTP it's easy to get within 10-15 ms of UTC, even with an
intermittent connection to the net. The NTP daemon measures the
clock's frequency error and slews it to compensate. Since most of the
error in a typical PC clock crystal is static frequency error, this
is remarkably effective.

NTP is native to the UNIX world (BSD, Linux). Don't know about support
for the Windows world.

Phil

115. COHERENT BPSK interface for EVM

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 18 Jul 1997 15:12:17 -0800
To: HFSIG@tapr.org, DSP@tapr.org
Message-ID: <199707172212.PAA02723@peak.org>
Rendered body type: text/plain

Hi all,

I have uploaded a package containing the DSP source code and notes on how to
use the EVM into a front end for the VE2IQ, "COHERENT" BPSK program. This is
a somewhat advanced project, so see it as a challenge - OK?

Please look for COHEVM1.ZIP in the TAPR DSP, recent_uploads area.

ftp://ftp.tapr.org/tapr/SIG/DSP/recent_uploads/COHEVM1.ZIP

It will appear there as soon as system admin (Tim Please) has moved it there.

Have fun and enjoy!

--Johan,

116. unsubscribe hfsig

From: Ed Kaprocki <kaprocki@iag.net>
Date: Thu, 17 Jul 1997 21:15:19 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BC92F6.8AE4ECC0@d203.aspg.fl.iag.net>
Rendered body type: text/plain

[Empty message body]

117. Re: [HFSIG:2323] Re: PC modem implementation (was: Serial

From: Lyle Johnson <lylej@azstarnet.com>
Date: Thu, 17 Jul 1997 18:17:59 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19970717180631.006feb94@pop.azstarnet.com>
Rendered body type: text/plain

At 03:10 PM 7/17/97 -0500, you wrote:
>> >If someone has another idea, please contact me :)
>>
>> Tha TAPR TAC project uses GPS to provide a clock. The follow-on (there we
>> go with dedicated hardware again, why not just use a virtual clock :-)
>> project will be the TOC< which is a disciplined 10 MHz oscillator sync'ed
>> to GPS to provide a reference accurate to about 1 part 10^9.
>>
>> Cheers,
>>
>> Lyle
>>
>What?! Geez! I don't believe you wrote that. This kind of accuracy is quite
>unnecessary and pointless and and and, quite honestly words fail me.

Antony,

The TOC isn't designed for HF modem use -- it is simply being designed as a
very accurate reference oscillator for whatever one wishes to use it for.

Why do I need it? I don't.

Will I get one and use it anyway? You bet!

Why? It's a hobby :-)

Cheers,

Lyle

118. Re: Error Condition Re: Re: [HFSIG:2314] 64 tone, 10 baud, BPSK modem

From: RAYMOND CZYZEWSKI <wa2sei@nac.net>
Date: Thu, 17 Jul 1997 23:52:39 -0400
To: hfsig@tapr.org
Message-ID: <33CEE887.1246@nac.net>
Rendered body type: text/plain

hfsig@tapr.org wrote:
>
> We are sorry, but this system sensed the following request which may have been
> inadvertedly sent to this list:
>
> UNSUBSCRIBE HFSIG
>

> If your posting was intentional, please accept our apologies and resend your
> mail message, making sure you do not include anything that may look like a
> request in the first line of the body of the actual message. If this was
> indeed a request please resend it to listproc@tapr.org
> Your entire message
> is copied below.
>
> ---
> unsubscribe hfsig

UNSUBSCRIBE HFSIG
PLEASE!!!
!!!!!!!!!!!!!!!!!!!!!!!!!!!
THANK YOU

119. Re: Error Condition Re: Re: [HFSIG:2317] Re: PC modem implementation (was: Serial

From: RAYMOND CZYZEWSKI <wa2sei@nac.net>
Date: Thu, 17 Jul 1997 23:53:54 -0400
To: hfsig@tapr.org
Message-ID: <33CEE8D2.62C3@nac.net>
Rendered body type: text/plain

hfsig@tapr.org wrote:
>
> We are sorry, but this system sensed the following request which may have been
> inadvertedly sent to this list:
>
> UNSCRIBS
>
> If your posting was intentional, please accept our apologies and resend your
> mail message, making sure you do not include anything that may look like a
> request in the first line of the body of the actual message. If this was
> indeed a request please resend it to listproc@tapr.org
> Your entire message
> is copied below.
>
> ---
> unscribs hfsig

120. Re: Error Condition Re: Re: [HFSIG:2314] 64 tone, 10 baud, BPSK modem

From: RAYMOND CZYZEWSKI <wa2sei@nac.net>
Date: Thu, 17 Jul 1997 23:54:24 -0400
To: hfsig@tapr.org
Message-ID: <33CEE8F0.952@nac.net>
Rendered body type: text/plain

hfsig@tapr.org wrote:
>
> We are sorry, but this system sensed the following request which may have been
> inadvertedly sent to this list:
>
> UNSUBSCRIBE HFSIG
>
> If your posting was intentional, please accept our apologies and resend your
> mail message, making sure you do not include anything that may look like a
> request in the first line of the body of the actual message. If this was
> indeed a request please resend it to listproc@tapr.org
> Your entire message
> is copied below.
>
> ---
> unsubscribe hfsig

121. Re: 64 tone, 10 baud, BPSK modem

From: Robert Schneider <rs@uni-wuppertal.de>
Date: Fri, 18 Jul 1997 11:16:04 +0200
To: hfsig@tapr.org
Message-ID: <33CF3454.15FB@uni-wuppertal.de>
Rendered body type: text/plain

Pawel,
where can I find your code?

Regards
Robert (db1jj)

122. RE: [HFSIG:2332] Re: 64 tone, 10 baud, BPSK modem

From: jalocha@axdele.cern.ch
Date: Fri, 18 Jul 1997 11:42:40 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199707180942.LAA02288@dxmint.cern.ch>
Rendered body type: text/plain

Hello,

>where can I find your code?

Please tell me your e-mail (mine: jalocha@axdele.cern.ch)
and I will send you the first version.
The code will be on the TAPR server after Johan checks that the modem
actually performs on the air :-)
And I am still making minor modifications...

Pawel

123. [HFSIG:2324] unsubscribe hfsig

From: Wolf Baron <wbaron.hondutel.hn@hondutel.hn>
Date: Fri, 18 Jul 1997 11:28:31
To: hfsig@tapr.org
Message-ID: <3.0.2.16.19970718112831.33df3cbe@hondutel.hn>
Rendered body type: text/plain

>Date: Thu, 17 Jul 1997 16:27:43 -0500 (CDT)
>Reply-To: hfsig@tapr.org
>Originator: hfsig@tapr.org
>Sender: hfsig@tapr.org
>From: Wolf Baron <wbaron.hondutel.hn@hondutel.hn>
>To: hfsig@tapr.org
>Subject: [HFSIG:2324] unsubscribe hfsig
>X-Comment: Tucson Amateur Packet Radio HF Special Interest Group
>X-Mailer: QUALCOMM Windows Eudora Light Version 3.0.2 (16)
>
>please don`t send more mail! 73 - Wolf
>
>

124. Re: [HFSIG:2314] 64 tone, 10 baud, BPSK modem

From: "Frank Knoll" <fknoll@ibm.net>
Date: Sat, 19 Jul 97 01:19:24 +0100
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <199707182320.XAA65580@pop01.ny.us.ibm.net>
Rendered body type: text/plain

On Thu, 17 Jul 1997 07:48:20 -0500 (CDT), jalocha@axdele.cern.ch (Pawel) wrote:

>Hi all,
>
>I have just made a working code for the 64-tone, 10 baud BPSK modem.
>
>It sends 64 tones, spaced by 15 Hz, allocated in the 500-1500 Hz audio range.
>Each tone is DBPSK modulated at 10 baud thus we get 64-bit data blocks
>and 10 blocks per second. User data is organized as 7-bit ASCII character
>stream. Each character is encoded with Walsh functions onto a 64-bit codeword.
>The codewords are then interleaved which spreads each character over about
>3 seconds. Overall user data rate is thus 10 7-bit ASCII characters per second.
>
>The modem audio input/output design is based on 8000 Hz sampling
>and an anti-alias filter which passes no more than 1600 Hz around the
>1000 Hz point. The data is down sampled by 5 (8000/5 = 1600 Hz)
>and then follows a 512 point FFT with sliding window.
>
>The demodulator is a pure phase approach: after the FFT I turn I/Q vectors
>into phases and all following analysis is done on these phases.
>I did so due to memory constraints of the EVM56K (I need to keep
>quite some history of the FFT slices) and because it made the code
>a bit simpler (you only do phase addition, subtraction, averaging, etc.).
>The demodulator computes soft decisions and the decoder is done with the
>fast (soft) Hadamard transform.
>
>The receiver can lock within about +/- 50 Hz and the lock range can be made
>larger in principle. There are no dedicated tuning/sync. tones - the receiver
>is able to lock within 5-10 seconds onto an ongoing transmition.
>After the lock the receiver tracks the frequency drift (up to some 0.05Hz/sec
>I think) and the timing drift too.
>
>The user interface is RTTY-like: Ctrl-T turns the transmitter on,

>Ctrl-R turns it off. Any character sent to the DSP is transmitted
>(if the transmitter is off, the data is buffered),
>whatever the receiver captures is sent to the terminal
>(after a signal is acquired).
>
>Who will try it first on the air ? :-)
>
>Pawel
>
>

Hello Pawel,

please explain how your 10baud BPSK modem makes
the frequency + timing sync.

Frank

best regards
Frank Knoll

==
* Frank Knoll (DL2FCH)
* Spessartstr. 12
* D-63571 Gelnhausen-Meerholz
* Germany
* Tel.: +49 6051 969116
* E-Mail: fknoll@ibm.net   or dl2fch@amsat.org
==

125. Re: [HFSIG:2314] 64 tone, 10 baud, BPSK modem

From: zs6awk@global.co.za (Danie Brynard)

Date: Sat, 19 Jul 1997 09:06:13 +0200
To: hfsig@tapr.org
Message-ID: <199707190704.JAA09915@mail.global.co.za>
Rendered body type: text/plain

Sounds like good work Pawel ! I would like to try it here locally. I guess
it will work much better on UHF hey ?

My email is: zs6awk@global.co.za

Danie zs6awk

>Hi all,
>
>I have just made a working code for the 64-tone, 10 baud BPSK modem.
>
>It sends 64 tones, spaced by 15 Hz, allocated in the 500-1500 Hz audio range.
>Each tone is DBPSK modulated at 10 baud thus we get 64-bit data blocks
>and 10 blocks per second. User data is organized as 7-bit ASCII character
>stream. Each character is encoded with Walsh functions onto a 64-bit codeword.
>The codewords are then interleaved which spreads each character over about
>3 seconds. Overall user data rate is thus 10 7-bit ASCII characters per second.
>
>The modem audio input/output design is based on 8000 Hz sampling
>and an anti-alias filter which passes no more than 1600 Hz around the
>1000 Hz point. The data is down sampled by 5 (8000/5 = 1600 Hz)
>and then follows a 512 point FFT with sliding window.
>
>The demodulator is a pure phase approach: after the FFT I turn I/Q vectors
>into phases and all following analysis is done on these phases.
>I did so due to memory constraints of the EVM56K (I need to keep
>quite some history of the FFT slices) and because it made the code
>a bit simpler (you only do phase addition, subtraction, averaging, etc.).
>The demodulator computes soft decisions and the decoder is done with the
>fast (soft) Hadamard transform.
>
>The receiver can lock within about +/- 50 Hz and the lock range can be made
>larger in principle. There are no dedicated tuning/sync. tones - the receiver
>is able to lock within 5-10 seconds onto an ongoing transmition.
>After the lock the receiver tracks the frequency drift (up to some 0.05Hz/sec
>I think) and the timing drift too.
>
>The user interface is RTTY-like: Ctrl-T turns the transmitter on,
>Ctrl-R turns it off. Any character sent to the DSP is transmitted
>(if the transmitter is off, the data is buffered),
>whatever the receiver captures is sent to the terminal
>(after a signal is acquired).

>
>Who will try it first on the air ? :-)
>
>Pawel
>
>

++
Danie Brynard   Email: zs6awk@global.co.za
PO Box 15133,Lyttelton,0140,South Africa
GRID LOC: KG44CE,   Tel/Fax: (2712) 6640330
++

126. RE: [HFSIG:2335] Re: 64 tone, 10 baud, BPSK modem

From: jalocha@axdele.cern.ch
Date: Sat, 19 Jul 1997 12:26:06 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199707191026.MAA27736@dxmint.cern.ch>
Rendered body type: text/plain

Hello Frank,

>please explain how your 10baud BPSK modem makes
>the frequency + timing sync.

In simplest words I can say that the receiver when in idle verifies the signal
hipothesis at each FFT bin (the tones are spaced by 5 bins) and at four
different times within the symbol period. When a match to the tone's
pattern is found and if it is good enough the receiver corrects the frequency
offset and swwitches to the tracking and decoding phase - there is refines
the tuning and timing.
The signal hipothesis is based on taking the phase difference between
FFT samples spaced by the symbol time and computing the average
and deviation across all tones. For a zero signal the average
is the frequency offset and the deviation reflects the signal/noise.
The task takes 20-25% CPU on the 40 MHz EVM56K thus in principle
one could do a more fine search grid but you need some more memory for this
to keep the history of FFT slices and do longer FFTs (if you want to refine
the grid in frequency).

Pawel

127. Re: [HFSIG:2337] Re: 64 tone, 10 baud, BPSK modem

From: zs6awk@global.co.za (Danie Brynard)
Date: Sun, 20 Jul 1997 08:01:37 +0200
To: hfsig@tapr.org
Message-ID: <199707200600.IAA05454@mail.global.co.za>
Rendered body type: text/plain

Pawel how can we classify the modem, is it an OFDM type ?

>Hello Frank,
>
>>please explain how your 10baud BPSK modem makes
>>the frequency + timing sync.
>
>In simplest words I can say that the receiver when in idle verifies the signal
>hipothesis at each FFT bin (the tones are spaced by 5 bins) and at four
>different times within the symbol period. When a match to the tone's
>pattern is found and if it is good enough the receiver corrects the frequency
>offset and swwitches to the tracking and decoding phase - there is refines
>the tuning and timing.
>The signal hipothesis is based on taking the phase difference between
>FFT samples spaced by the symbol time and computing the average
>and deviation across all tones. For a zero signal the average
>is the frequency offset and the deviation reflects the signal/noise.
>The task takes 20-25% CPU on the 40 MHz EVM56K thus in principle
>one could do a more fine search grid but you need some more memory for this
>to keep the history of FFT slices and do longer FFTs (if you want to refine
>the grid in frequency).
>
>Pawel
>
>

++
Danie Brynard   Email: zs6awk@global.co.za
PO Box 15133,Lyttelton,0140,South Africa
GRID LOC: KG44CE,   Tel/Fax: (2712) 6640330

++

128. Re: [HFSIG:2328] Re: PC modem implementation (was: Serial

From: Antony Martin <anthony.martin@gecm.com>
Date: Mon, 21 Jul 1997 13:37:20 +0100
To: hfsig@tapr.org
Message-ID: <970721133720.6666@oas01.gmcl.co.uk>
Rendered body type: text/plain

> Antony,
>
> The TOC isn't designed for HF modem use -- it is simply being designed as a
> very accurate reference oscillator for whatever one wishes to use it for.

> Lyle
>
My apologies - I was ignorant of the entirely unrelated nature of the
TAC project. An accurate lab standard sounds pretty useful to me.

Anthony.

Perhaps people could actually change the Subject field when the subject
changes... :-)

129. Re: [HFSIG:2277] Re: Serial versus parallel, mainly parallel

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 22 Jul 1997 21:36:52 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707230436.VAA19423@servo.qualcomm.com>
Rendered body type: text/plain

>due to my ignorance. So Phil, could you give us pointers to the necessary
>know-how on doing DSP in a Pentium? If you have time a simple HOWTO on

Fundamentally, it's really no different than doing DSP on any other
hardware platform. The algorithms are the same, only the hardware you
run them on is different. Sure, there are programming tricks to be
used if you really want to get the most out of a particular chip; see
my page

<http://people.qualcomm.com/karn/ham.html>

for some particular examples of DSP on a Pentium.

Phil

130. HF BSPSK BEACON

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 24 Jul 1997 08:27:10 -0800
To: HFSIG@tapr.org
Message-ID: <199707231527.IAA15179@peak.org>
Rendered body type: text/plain

Hi folks,

In order to explore trans-Atlantic BPSK communications there is exciting
news for those interested in HF BPSK work. Perhaps you would like to
participate?

Dave, AA1A will be running a beacon on 14081 (carrier) this weekend,
Saturday at 1600Z. It is the COHERENT format. Dave is located in
Massachusetts and will be beaming towards Europe.

Those that want to run COHERENT, obtain my EVM software package from:

ftp://ftp.tapr.org/tapr/SIG/dsp/recent_uploads/cohevm11.zip

There are some instructions on how to set it up and also where to obtain the
rest of the software. I have been using this software successfully for weak
signal work.

Hope you find this of interest.

73's

Johan, KC7WW

131. SQPSK

From: "Bergzen Hakan, KR" <hakan.bergzen@enator.se>
Date: Thu, 24 Jul 1997 09:54:00 +0100
To: hfsig <hfsig@tapr.org>
Message-ID: <33DC9664@noak>
Rendered body type: text/plain

Phil Karn wrote:
>...snip. Sure, there are programming tricks to be
>used if you really want to get the most out of a particular chip; see
>my page
>http://people.qualcomm.com/karn/ham.html
>for some particular examples of DSP on a Pentium.

Which I did, and found to be very interesting pages. Phil, could you please clarify whether one thing, though. You use what you describe as Staggered QPSK (SQPSK). I have not heard this expression before. It seems to me to be very similar to Offset-Keyed QPSK (OKQPSK). Are they identical?

Regarding digitised voice: I had this idea about possible reduction in CELP data rate. Let's say that you make a Huffman table of the codebook indexes for your particular voice. Upon initialisation of a digitised voice link the two operators exchange Huffman tables. Do you think this would reduce the overall data rate (or is it already included in the schemes perhaps)? If the operators normally speak to each other there would of course not be a need to exchange Huffman tables every time (merely an identification of which stored Huffman table to use).

Just curious,
Hakan

132. Re: [HFSIG:2343] SQPSK

From: Antony Martin <anthony.martin@gecm.com>
Date: Thu, 24 Jul 1997 12:54:28 +0100
To: hfsig@tapr.org
Message-ID: <970724125428.28880@oas01.gmcl.co.uk>
Rendered body type: text/plain

>
> Regarding digitised voice: I had this idea about possible reduction in CELP
> data rate. Let's say that you make a Huffman table of the codebook indexes
> for your particular voice. Upon initialisation of a digitised voice link the
> two operators exchange Huffman tables. Do you think this would reduce the
> overall data rate (or is it already included in the schemes perhaps)? If the
> operators normally speak to each other there would of course not be a need
> to exchange Huffman tables every time (merely an identification of which
> stored Huffman table to use).
>
> Just curious,
> Hakan
>
For realtime speech you need a guaranteed data rate that matches the
channel rate. Huffman coding gives a variable data rate.
The codebook already contains the most used entries anyway, and
possible choices for the codebook that were little used didn't
make it to the codebook. Hence huffman coding may not help.

Perhaps you have statistics on CELP codebook use for your voice
to contradict this?

If you want lower rate speech, see:
ftp://svr-ftp.eng.cam.ac.uk/comp.speech/coding/lpc10-1.0.tar.gz
which offers 2400b/s and rates down to 800b/s LPC10 coding.

As speech can tolerate high error rates, try not using any
FEC coding on your HF modem.

Anthony.

133. Re: [HFSIG:2343] SQPSK

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 24 Jul 1997 09:46:11 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707241646.JAA07256@servo.qualcomm.com>
Rendered body type: text/plain

>Which I did, and found to be very interesting pages. Phil, could you please
>clarify whether one thing, though. You use what you describe as Staggered
>QPSK (SQPSK). I have not heard this expression before. It seems to me to be

>very similar to Offset-Keyed QPSK (OKQPSK). Are they identical?

Yes, SQPSK == OK-QPSK. Two terms for the same thing.

Re digital voice encoding, that's not my field so I can't say how well
it would work.

Phil

134. Re: [HFSIG:2344] Re: SQPSK

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 24 Jul 1997 10:24:58 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707241724.KAA08373@servo.qualcomm.com>
Rendered body type: text/plain

>As speech can tolerate high error rates, try not using any
>FEC coding on your HF modem.

I seriously question this. *Uncompressed* speech can handle high error
rates because it has a lot of built-in redundancy. But speech coders
are specifically designed to remove as much of this redundancy as
possible. This makes the encoded bits far more fragile than the raw
uncompressed input. Hence you need FEC to protect them.

That's certainly been our experience with CDMA. We use a CELP-based
vocoder that is rather sensitive to errors, so we use convolutional
coding and add a CRC to each frame as it goes over the air. Better to
erase the frame entirely than hand it to the vocoder with uncorrected
errors. When that happened from time to time in our lab (usually due
to some bug), the decompressor generally went into the aptly-named
"Jimi Hendrix mode". Something you want to avoid in an operational
system!

Removing redundancy with source coding only to add it back in with
channel coding may seem like a Chinese fire drill, but it's the right
thing to do. (The Shannon Channel Separation Theorem says so). The
redundancy in speech is not very efficiently designed to deal with
channel errors, or at least it's much less efficient than the very
carefully designed redundancy in a forward error correcting code.

Phil

135. Re: [HFSIG:2346] Re: SQPSK

From: Brian Beezley <k6sti@n2.net>
Date: Thu, 24 Jul 1997 13:48:25 -0700
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19970724134838.11cfacbc@mail.n2.net>
Rendered body type: text/plain

At 12:55 PM 7/24/97 -0500, you wrote:
>>As speech can tolerate high error rates, try not using any
>>FEC coding on your HF modem.
>
>I seriously question this. *Uncompressed* speech can handle high error
>rates because it has a lot of built-in redundancy. But speech coders
>are specifically designed to remove as much of this redundancy as
>possible. This makes the encoded bits far more fragile than the raw
>uncompressed input. Hence you need FEC to protect them.
>
>That's certainly been our experience with CDMA. We use a CELP-based
>vocoder that is rather sensitive to errors, so we use convolutional
>coding and add a CRC to each frame as it goes over the air. Better to
>erase the frame entirely than hand it to the vocoder with uncorrected
>errors. When that happened from time to time in our lab (usually due
>to some bug), the decompressor generally went into the aptly-named
>"Jimi Hendrix mode". Something you want to avoid in an operational
>system!
>
>Removing redundancy with source coding only to add it back in with
>channel coding may seem like a Chinese fire drill, but it's the right
>thing to do. (The Shannon Channel Separation Theorem says so). The
>redundancy in speech is not very efficiently designed to deal with
>channel errors, or at least it's much less efficient than the very
>carefully designed redundancy in a forward error correcting code.
>
>Phil
>
>

Phil, I think it all depends on the speech-compression algorithm. When I
was doing 2400 and 4800 bps vocoders (not LPC-10 but something similar), we
used to do everything we could to obtain an ARQ-free channel. The ARQ delay
would trash the speech quality, but quite a few bits could be in error with
our algorithms without audible problems. However, your CELP probably is
much more efficient (for the reproduced speech quality) than the rather
primitive algorithms we used, so it can't tolerate even a few bad bits.
Speech coding is part black art, and speech-quality evaluation is entirely
subjective, so I think it's difficult to reliably generalize about these
matters.

What is the channel separation theorem?

Brian Beezley, K6STI
k6sti@n2.net

136. Re: [HFSIG:2347] Re: SQPSK

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 24 Jul 1997 16:36:43 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.970724163522.9891D-100000@www.kelly-afb.org>
Rendered body type: text/plain

Brian,

RE:ARQ...We discovered about the same thing with LCP-10.   -- Walt

```
    ========================================================================
    |        If I wanted "Windows", I'd knock a hole in my wall.           |
    ========================================================================
    |                                  |                                   |
    |                                  | The Micro$oft operating system    |
    | Walt DuBose - K5YFW              | didn't get as bad as it is        |
    | E-Mail k5yfw@www.kelly-afb.org   | overnight.  I has taken over 10   |
    | Business Telephone: (210)925-1171 | years of careful, calculated,    |
    |     Home Telephone: (210)696-3196 | development.                     |
    |                                  |                                   |
    ========================================================================
```

On Thu, 24 Jul 1997, Brian Beezley wrote:

> At 12:55 PM 7/24/97 -0500, you wrote:
> >>As speech can tolerate high error rates, try not using any
> >>FEC coding on your HF modem.
> >
> >I seriously question this. *Uncompressed* speech can handle high error
> >rates because it has a lot of built-in redundancy. But speech coders
> >are specifically designed to remove as much of this redundancy as
> >possible. This makes the encoded bits far more fragile than the raw
> >uncompressed input. Hence you need FEC to protect them.
> >
> >That's certainly been our experience with CDMA. We use a CELP-based
> >vocoder that is rather sensitive to errors, so we use convolutional
> >coding and add a CRC to each frame as it goes over the air. Better to
> >erase the frame entirely than hand it to the vocoder with uncorrected
> >errors. When that happened from time to time in our lab (usually due
> >to some bug), the decompressor generally went into the aptly-named
> >"Jimi Hendrix mode". Something you want to avoid in an operational
> >system!
> >
> >Removing redundancy with source coding only to add it back in with
> >channel coding may seem like a Chinese fire drill, but it's the right
> >thing to do. (The Shannon Channel Separation Theorem says so). The
> >redundancy in speech is not very efficiently designed to deal with
> >channel errors, or at least it's much less efficient than the very
> >carefully designed redundancy in a forward error correcting code.
> >
> >Phil
> >
> >
>
>
> Phil, I think it all depends on the speech-compression algorithm. When I
> was doing 2400 and 4800 bps vocoders (not LPC-10 but something similar), we
> used to do everything we could to obtain an ARQ-free channel. The ARQ delay
> would trash the speech quality, but quite a few bits could be in error with

> our algorithms without audible problems. However, your CELP probably is
> much more efficient (for the reproduced speech quality) than the rather
> primitive algorithms we used, so it can't tolerate even a few bad bits.
> Speech coding is part black art, and speech-quality evaluation is entirely
> subjective, so I think it's difficult to reliably generalize about these
> matters.
>
> What is the channel separation theorem?
>
>
> Brian Beezley, K6STI
> k6sti@n2.net
>
>

137. Re: [HFSIG:2346] Re: SQPSK

From: Dave Emery <die@pig.die.com>
Date: Fri, 25 Jul 1997 00:30:06 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199707250430.AAA01685@pig.die.com>
Rendered body type: text/plain

Phil Karn wrote :

> That's certainly been our experience with CDMA. We use a CELP-based
> vocoder that is rather sensitive to errors, so we use convolutional
> coding and add a CRC to each frame as it goes over the air. Better to
> erase the frame entirely than hand it to the vocoder with uncorrected
> errors. When that happened from time to time in our lab (usually due
> to some bug), the decompressor generally went into the aptly-named
> "Jimi Hendrix mode". Something you want to avoid in an operational
> system!
>

For what little it is worth, INMARSAT Aero H uses a hamming code for "the critical partial correllation coefficient bits" and a column parity scheme as well. This allows correction of a single bit error and detection of doubtle and multiple bit errors in most cases.

If a double bit error happens they use the previous 32 ms frame coefficients if they were good or correctable and shut off audio ("mute the codec") otherwise. If more than 3 errors happen they mute audio regardless. When they mute, they set to midrange some of the longer term predictors.

Partial correlation coefficients are used to derive the filter tap values for the receive vocal tract filter. I guess most of the rest of the stuff in the frame doesn't produce as striking a tweet...

The system error budget allows a BER of 10**-3 for the 10.5 kbs output of the Vitirbi in voice mode, so the basic vocoder must be pretty robust as far as corruption is concerned..

I forget what your stuff does for error detection on vocoder frames... I have the standard somewhere...

Dave

138. Re: [HFSIG:2349] Re: SQPSK

From: Dave Emery <die@pig.die.com>
Date: Fri, 25 Jul 1997 01:57:11 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199707250557.BAA02292@pig.die.com>
Rendered body type: text/plain

Dave Emery wrote :

>   I forget what your stuff does for error detection on vocoder
> frames... I have the standard somewhere...
>

SORRY, I intended that to be addressed to Phil... and didn't notice the unusual reply-to configuration...

Hope noone minds...

Dave

139. Re: SQPSK

From: "Bergzen Hakan, KR" <hakan.bergzen@enator.se>
Date: Fri, 25 Jul 1997 08:27:00 +0100
To: hfsig <hfsig@tapr.org>
Message-ID: <33DDD351@noak>
Rendered body type: text/plain

Anthony Martin wrote:
>For realtime speech you need a guaranteed data rate that matches the
>channel rate. Huffman coding gives a variable data rate.
>The codebook already contains the most used entries anyway, and
>possible choices for the codebook that were little used didn't
>make it to the codebook. Hence huffman coding may not help.

Yes, the idea would have to fulfil two basic requirements in order to be effective:
1. An individual is expected to use only a part of the possible codebook entries for most of the time
2. These most-popular codebook entries shall differ between individuals (otherwise skip the rest of the entries (and by the way, is that not what is done in reduced rate LPC10e schemes?))

Is it not sufficient if the maximum voice digitisation data rate matches the modem data rate? The intention would be to be able to reduce the data rate by some factor, thus not needing e.g. 2400 bps modems.Huffman pauses and voice pauses could be made better use of by allowing transfer of system or modem link information during those instances.

>Perhaps you have statistics on CELP codebook use for your voice
>to contradict this?
Sorry to say no. I have no real hands-on experience of voice digitisation schemes (although I hope that can be changed in the future :-)

What I envision is a portable PC with a soundblaster card controlling an HF radio probably via a black box. The PC will house the application MMI, the signalling protocols and FEC, and ideally also the modem (otherwise to be housed in the black box). Voice digitisation would be enabled via the use of embedded software like LPC10e (preferably in the form of already available www programs such as Speak Freely). Intranet integration via web browsers will enable in-house global direct access without the need for specific user terminal equipment. Even internet integration for forwarding store-and-forward based traffic (like email) could also easily be arranged. The ham community have already developed parts of these functionalities, but I am not aware of any complete system. It should of course not rely or use any proprietary schemes whatsoever (at least not the over-the-air signalling format).

Hakan

140. Re: [HFSIG:2350] Re: SQPSK

From: Brian Beezley <k6sti@n2.net>
Date: Fri, 25 Jul 1997 00:06:54 -0700
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19970725000706.197fdcc0@mail.n2.net>
Rendered body type: text/plain

At 12:58 AM 7/25/97 -0500, you wrote:
>Dave Emery wrote :
>
>>   I forget what your stuff does for error detection on vocoder
>> frames... I have the standard somewhere...
>>
>
>   SORRY, I intended that to be addressed to Phil... and didn't
>notice the unusual reply-to configuration...
>
>   Hope noone minds...
>
>   Dave
>
>
>

Same thing happened to my reply to Phil, Dave. The message popped out here unintentionally. I sure wish HFSIG didn't put itself in the reply address!

Brian Beezley, K6STI

k6sti@n2.net

141. Re: SQPSK

From: "Bergzen Hakan, KR" <hakan.bergzen@enator.se>
Date: Fri, 25 Jul 1997 11:11:00 +0100
To: hfsig <hfsig@tapr.org>
Message-ID: <33DDFA78@noak>
Rendered body type: text/plain

```
>>       SORRY,  I intended that to be addressed to Phil... and didn't
>>notice the unusual reply-to configuration...
>>
>>       Hope noone minds...
>
>Same thing happened to my reply to Phil, Dave.  The message popped out here
>unintentionally.  I sure wish HFSIG didn't put itself in the reply address!
>

Even though your comments were not intended for the rest of us, I certainly
found them interesting to read. Since this is a discussion group, then why
mailing such
informative replies to one person only?

Just my thoughts...

Hakan
```

142. Re: [HFSIG:2353] Speech coding (was SQPSK)

From: Antony Martin <anthony.martin@gecm.com>
Date: Fri, 25 Jul 1997 15:46:29 +0100
To: hfsig@tapr.org
Message-ID: <970725154629.17092@oas01.gmcl.co.uk>
Rendered body type: text/plain

```
A few points to bear in mind:

As all bits in speech don't cause the same effect when an error hits
them, your coding scheme should be designed to be more robust for the
most significant bits. Coded speech has ms bits and ls bits like
ordinary speech. The ls bits are not worth correcting; the ms bits are.

LPC10e (the mil standard) includes FEC of the higher bits in unvoiced
sounds and smoothing of pitch and reflection coefficients which
will resist large unexpected changes caused by channel errors.
It also measures channel BER to guide it's smoothing process.
It is also possible to use signal quality information to do
bad-frame replacement.

Internet speech applications such as Speak Freely are designed for
variable-rate error-free channels.

Huffman coding can actually increases the data rate if the channel
data does not conform to the expected symbol distribution. No
lossy compression algorithm will compress all possible files. A
small set of files are compressed at the expense of making all other
possible files larger. Most lossless coding scemes perform badly
in the presence of a small amount of added noise in the data, eg
like in analogue data. This is because real noise in non-compressible.

Data compression schemes can cause error extension. ie. a 1 bit
error can turn into lots of errors. In some cases can make the
data unrecoverable, or cause loss of sync with the code.

Variable symbol-size coding should _never_ be used on an errored link.

The 600, 800 and 1200 bits/sec versions of LPC are vector-quantised.
In the case of 600, one of 3 vector codebooks is used for each
frame.

Why the preoccupation with user interfaces for DSP applications;
you turn it on and it just works, doesn't it?

What things are needed on a user interface apart from data in
and data out?
```

143. Re: [HFSIG:2352] Re: SQPSK

From: Dave Emery <die@pig.die.com>
Date: Fri, 25 Jul 1997 14:06:54 -0400 (EDT)

To: hfsig@tapr.org
Message-ID: <199707251806.OAA05053@pig.die.com>
Rendered body type: text/plain

Brian Beezley wrote :

>
>
> Same thing happened to my reply to Phil, Dave. The message popped out here
> unintentionally. I sure wish HFSIG didn't put itself in the reply address!

        It is a configuration option for listserv or majordomo, and most
everyone else discovers the problems it causes and changes it
to put the senders address in...

                                                        Dave

144. Re: Speech coding (was SQPSK)

From: "Bergzen Hakan, KR" <hakan.bergzen@enator.se>
Date: Fri, 25 Jul 1997 22:22:00 +0100
To: hfsig <hfsig@tapr.org>
Message-ID: <33DE9687@noak>
Rendered body type: text/plain

Good points, Andrew. I guess I will have to get the LPC10e (FS-1015)
standard and learn it before commenting or trying to change it :-)

>Internet speech applications such as Speak Freely are designed for
>variable-rate error-free channels.

I actually meant using Speak Freely on the intranet only , and then convert
it either to analogue (via the intranet web servers soundblaster card) or to
LPC10e before feeding it to the radio connected to the intranet web server
(but it did not came out that clear in my Swedish-to-English conversion :-)

>Why the preoccupation with user interfaces for DSP applications;
>you turn it on and it just works, doesn't it? What things are needed on
>a user interface apart from data in and data out?

On link level? Nothing I guess, but as an application user I want buttons
and knobs and speakers and leds and switches and gauges and... :-) In short,
the world fancies fancy interfaces even if they are only simulated on a PC
screen.
E.g. I for one liked WordPerfect for DOS, but then everyone at work said
that it was impossible and old-fashioned to work with and I was forced to
change to Word for Windows. Nowadays I cannot make the nice graphs I managed
in DrawPerfect, and I still miss the WP code window (or whatever it is
called in the english version) that put you in command of the document
production process (instead of some Word wizard that never does what you
want it to).

Hakan

145. Re: [HFSIG:2352] Re: SQPSK

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 25 Jul 1997 15:42:42 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707252242.PAA26044@servo.qualcomm.com>
Rendered body type: text/plain

>Same thing happened to my reply to Phil, Dave. The message popped out here
>unintentionally. I sure wish HFSIG didn't put itself in the reply address!

Long time readers of this list will remember that I've complained
repeatedly and insistently about the broken behavior of the list
server, such as the bogus Reply-To: header, the deletion of other
recipients from the To: and CC.lists, and the munging of the Subject
line.

But, sadly, the maintainer seems to consider these bugs to be features.
And they've spread to other lists as well. Sigh.

Phil

146. Re: [HFSIG:2354] Re: Speech coding (was SQPSK)

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 25 Jul 1997 18:24:07 -0700 (PDT)
To: hfsig@tapr.org

Message-ID: <199707260124.SAA27213@servo.qualcomm.com>
Rendered body type: text/plain

>As all bits in speech don't cause the same effect when an error hits
>them, your coding scheme should be designed to be more robust for the
>most significant bits. Coded speech has ms bits and ls bits like
>ordinary speech. The ls bits are not worth correcting; the ms bits are.

The fact that this is true for many current speech coders says that we
still have a ways to go yet in making them optimal. An ideal speech
coder would produce bits that are *all* equally valuable. In fact,
changing even one bit in the output of an ideal encoder would change
the resulting reconstructed speech into a different but equally
plausible, grammatically correct sentence! Hence the increasing
importance of error control as source coding gets better.

Phil

147. Re: [HFSIG:2355] Re: SQPSK

From: John Koster <w9ddd@tapr.org>
Date: Fri, 25 Jul 1997 21:21:26 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <Pine.BSI.3.91.970725211558.14213B-100000@tapr.org>
Rendered body type: text/plain

Gee, guess I'm missing something. What's the big deal? My mail program
asks me if I want to reply to the list or the originator (default set to
originator). If I goof and select the wrong one, I can still go back and
edit the To: field manually.

Also, unless your response is very personal, most of the time comments
are of general interest and we'd all like to read them.

John, w9ddd@tapr.org

On Fri, 25 Jul 1997, Dave Emery wrote:

> Brian Beezley wrote :
>
> >
> >
> > > Same thing happened to my reply to Phil, Dave. The message popped out here
> > > unintentionally. I sure wish HFSIG didn't put itself in the reply address!
>
>   It is a configuration option for listserv or majordomo, and most
> everyone else discovers the problems it causes and changes it
> to put the senders address in...
>
>
>   Dave
>
>
>

148. Speech & text redundancy

From: Marijan Miletic <s56a@s55tcp.ampr.org>
Date: Sat, 26 Jul 97 06:42:05 EDT
To: hfsig@tapr.org
Message-ID: <27368@s55tcp.ampr.org>
Rendered body type: text/plain

While it is well known and thorougly discussed on this reflector that speech
has a lot of redundancy, it seems to me that the same has become with a text!
Walt, K5YFW wrote a single line sentence replying K6STI but then added 11 lines
of his ID and another 50 lines of qquotes.
And he is not even using M$ software :-)

73 de Mario, S56A, N1YU.

149. unsubscribe hfsig

From: "Robson Ribeiro da Silva" <rrdsilva@sti.com.br>
Date: Fri, 25 Jul 1997 16:19:58 -0300
To: <hfsig@tapr.org>
Message-ID: <199707261635.NAA02094@netra01.origin.com.br>
Rendered body type: text/plain

[Empty message body]

150. Re: [HFSIG:2341] HF BSPSK BEACON

From: zs6awk@global.co.za (Danie Brynard)
Date: Sat, 26 Jul 1997 19:28:45 +0200
To: hfsig@tapr.org
Message-ID: <199707261727.TAA15815@mail.global.co.za>
Rendered body type: text/plain

Johan is 1600Z for Oregon ? What is the UTC time ?

Danie zs6awk

>Hi folks,
>
>In order to explore trans-Atlantic BPSK communications there is exciting
>news for those interested in HF BPSK work. Perhaps you would like to
>participate?
>
>Dave, AA1A will be running a beacon on 14081 (carrier) this weekend,
>Saturday at 1600Z. It is the COHERENT format. Dave is located in
>Massachusetts and will be beaming towards Europe.
>
>Those that want to run COHERENT, obtain my EVM software package from:
>
>ftp://ftp.tapr.org/tapr/SIG/dsp/recent_uploads/cohevm11.zip
>
>There are some instructions on how to set it up and also where to obtain the
>rest of the software. I have been using this software successfully for weak
>signal work.
>
>Hope you find this of interest.
>
>73's
>
>Johan, KC7WW
>
>

++
Danie Brynard   Email: zs6awk@global.co.za
PO Box 15133,Lyttelton,0140,South Africa
GRID LOC: KG44CE,   Tel/Fax: (2712) 6640330
++

151. [HFSIG:2357] Re: SQPSK

From: kl7ev@actis.polarnet.com
Date: 26 Jul 1997 13:02:20 -0800 (AST)
To: hfsig@tapr.org
Message-ID: <199707262159.QAA09874@tapr.org>
Rendered body type: text/plain

HF>>Same thing happened to my reply to Phil, Dave. The message popped out here
HF>>unintentionally. I sure wish HFSIG didn't put itself in the reply address!

HF>Long time readers of this list will remember that I've complained
HF>repeatedly and insistently about the broken behavior of the list
HF>server, such as the bogus Reply-To: header, the deletion of other
HF>recipients from the To: and CC.lists, and the munging of the Subject
HF>line.

HF>But, sadly, the maintainer seems to consider these bugs to be features.
HF>And they've spread to other lists as well. Sigh.

HF>Phil

I must agree, however I thought it was just me or the way I get my mail
delivered. I think that however if I selected the "Digest" mode, that
it will retain the originators address. Ciao,

       - Greg -
----
kl7ev
----

152. Re: [HFSIG:2342] Message not deliverable

From: nt1g@bitwise.net (Skip Flem)
Date: Sun, 27 Jul 1997 09:58:55
To: hfsig@tapr.org
Message-ID: <27199755.Amiga@mailbox.bitwise.net>

Rendered body type: text/plain

> Work:SETUP/IBrowser2/mailout/1997.07.27.001.AM

> The following is an attached File item from cc:Mail. It contains
> information that had to be encoded to ensure successful transmission
> through various mail systems. To decode the file use the UUDECODE
> program.
> ---------- Cut Here ----------
>
> The following is an attached File item from cc:Mail. It contains
> information that had to be encoded to ensure successful transmission
> through various mail systems. To decode the file use the UUDECODE
> program.
> ---------- Cut Here ----------
>
>
>
>

--
Skip Flem, NT1G <nt1g@bitwise.net> Hull, Massachusetts 02045 U.S.A.

153. Re:[HFSIG:2342] Message not deliverable

From: forrerj@peak.org (Johan Forrer)
Date: Sun, 27 Jul 1997 11:17:36 -0800
To: hfsig@tapr.org
Message-ID: <199707271817.LAA18281@peak.org>
Rendered body type: text/plain

Hi,

Does anyone know what these 33 parts of a huge UUENCODED file contain and
who uploaded it to this list?
I suspect it was done in error or automatically by a server demon as I
cannot imagine anyone doing this intentionally.

Unless otherwise notified please just trash it without bothering to look at it.

Thanks,

Johan

154. Re: [HFSIG:2365] Re:Message not deliverable

From: Heikki Hannikainen <hessu@pspt.fi>
Date: Sun, 27 Jul 1997 21:53:19 +0300 (EEST)
To: hfsig@tapr.org
Message-ID: <Pine.LNX.3.96.970727214305.3524A-100000@zone.pspt.fi>
Rendered body type: text/plain

On Sun, 27 Jul 1997, Johan Forrer wrote:

> Does anyone know what these 33 parts of a huge UUENCODED file contain and
> who uploaded it to this list?

  It is a bounce of the MIME attached ZIP compressed GIF images of a US
patent some moron sent to the mailing list a while back. The simpleminded
cc:Mail gateway couldn't handle it (because of the size, or the MIME
attachments). The gateway doesn't know about MIME, and decided to uuencode
the contents to ensure the bounce would get through... and sent the bounce
to the mailing list. Let's hope the bounce doesn't start looping.

> I suspect it was done in error or automatically by a server demon as I
> cannot imagine anyone doing this intentionally.

  Uh-oh. Someone did send the very same images on the list before....

 < Heikki Hannikainen   <> ax.25:   oh7lzb@oh7rba.fin.eu   >
 < Internet: hessu@pspt.fi <> Amprnet: oh7lzb@oh7rba.ampr.org >

155. rtty

From: "Chris Terwilliger" <a229aa@email.mot.com>
Date: Mon, 28 Jul 1997 09:31:27 -0000
To: "HF SIG" <hfsig@tapr.org>
Message-ID: <9707281631.AA16342@mogate.sps.mot.com>
Rendered body type: text/plain

Could someone point me to sources or examples of RTTY decode software?
Someone in a previous thread got me to thinking about seeing what my Sun
IPX audio hardware / SPARC processor can really do and I'd like to start
with something simple like rtty.

tnx

/// Chris Terwilliger
/// Motorola
/// Phoenix Applied Research Center
/// a229aa@email.mot.com

156. Re: [HFSIG:2358] Re: Speech coding (was SQPSK)

From: Antony Martin <anthony.martin@gecm.com>
Date: Mon, 28 Jul 1997 19:07:53 +0100
To: hfsig@tapr.org
Message-ID: <970728190753.1246@oas01.gmcl.co.uk>
Rendered body type: text/plain

> >As all bits in speech don't cause the same effect when an error hits
> >them, your coding scheme should be designed to be more robust for the
> >most significant bits. Coded speech has ms bits and ls bits like
> >ordinary speech. The ls bits are not worth correcting; the ms bits are.
>
> The fact that this is true for many current speech coders says that we
> still have a ways to go yet in making them optimal. An ideal speech
> coder would produce bits that are *all* equally valuable. In fact,
> changing even one bit in the output of an ideal encoder would change
> the resulting reconstructed speech into a different but equally
> plausible, grammatically correct sentence! Hence the increasing
> importance of error control as source coding gets better.
>
> Phil
>
I'm inclined to agree with you that all bits should be equally
significant. But I don't think this is a necessary result of
perfect compression. But it is a requirement of good ECC that
all symbols be equiprobable and equally significant.

The basic problem is that it's impossible to code analogue values
without using a set of bits of different weights.

Correction: it's possible to encode analogue values into a set of
bits of arbitrarily close to equal weights, but not in bounded time.

Looking it up in a random lookup table would appear to solve
the weighting problem, but I'm not so sure, as the error
distribution doesn't seem right.

When an analogue signal is passed thru a digital comms channel
subject to AWGN, you should get the signal plus AWGN out the other
end, hopefully with more SNR as a result of the coding.

But the question is: A bit error is equally significant to whom?
To a computer all bits are created equal. To a human being, some
information is more important than other information. It's also
the receiver who determines if it's important but the sender that
compresses it.

In many cases, information has a shelf-life, and time is in itself
information, so perfect compression is impossible, although with
increases in processing speed, compression will become closer
to 'perfect', whatever that is for lossy compression. But for the
time being we must use what is 'optimal' using the bounded memory,
processing time, delay, etc. that is acceptable in each application.

As you suggest, future coders may represent all conceivable sentences
in say a 200 bit symbol. Furthermore, all sentences with the same
meaning in all languages will have the same representation. One
day maybe there will be compression algorithms that will remove
entire sentences as redundant or of no information to the receiver.
In that class would probably come jokes, comments about the weather,
all adverts and spam, all sport, sit-coms soaps and gameshows, and
almost anything said by a celebrity or politician. Pity It'll never
be in my lifetime.

And probably most Ham radio traffic! Information gained from a
link about the propagation conditions or the performance of the
equipment would suggest that the information content had increased

as a result of transmission. This would be totally impossible if
there is no redundancy left and all bits were fully utilised.

Personally I prefer a bit of redundancy as being human I tend to
forget, so a bit of unnecessary duplication isn't a bad thing.

'nuff of these ramblings

For now you've got LPC, CELP or nothing. Or delta mod. All bits
are equal in delta mod.

Anthony

***
You can experience my very good internet compression algorithm
free of charge. It achieves very high lossy compression ratios
of the order of 10^6:1 For every 1Kbytes transferred, there's
at least 1Gigabytes of eliminated redundant information. The
algorithm is as follows: 'Don't request anything you don't need'.
:-)
***

157. Re: [HFSIG:2361] unsubscribe hfsig

From: Monte.Tremont@nextel.com (Monte Tremont)
Date: Mon, 28 Jul 1997 14:28:09 -0400
To: hfsig@tapr.org, "Robson Ribeiro da Silva" <rrdsilva@sti.com.br>
Message-ID: <00034468.@nextel.com>
Rendered body type: text/plain

[Empty message body]

158. Re: [HFSIG:2368] Re: Speech coding (was SQPSK)

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 28 Jul 1997 20:22:35 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707290322.UAA00014@servo.qualcomm.com>
Rendered body type: text/plain

>The basic problem is that it's impossible to code analogue values
>without using a set of bits of different weights.

What about delta mod? You mention it later on, where you say that
all the bits are equal.

>When an analogue signal is passed thru a digital comms channel
>subject to AWGN, you should get the signal plus AWGN out the other
>end, hopefully with more SNR as a result of the coding.

Why? Real digital voice systems invariably give highly non-gaussian
noise outputs (assuming they fail at all) even when the channel is
gaussian. Even simple FEC schemes like the Viterbi decoder are well
known to produce errors in bursts when run on the AWGN channel. This
is as implied by coding theory. What's so terrible about that?

I tend to agree with the rest of your message, especially the stuff
about the content-free nature of much of ham radio and USENET...

Phil

159. List Behavior [Was Re: SQPSK]

From: Barry McLarnon VE3JF <bm@lynx.ve3jf.ampr.org>
Date: Tue, 29 Jul 1997 04:21:29 +0000 (GMT)
To: hfsig@tapr.org
Message-ID: <199707290421.EAA16202@lynx.ve3jf.ampr.org>
Rendered body type: text/plain

Phil Karn <karn@qualcomm.com> wrote:
> >Same thing happened to my reply to Phil, Dave. The message popped out here
> >unintentionally. I sure wish HFSIG didn't put itself in the reply address!
>
> Long time readers of this list will remember that I've complained
> repeatedly and insistently about the broken behavior of the list
> server, such as the bogus Reply-To: header, the deletion of other
> recipients from the To: and CC.lists, and the munging of the Subject
> line.
>
> But, sadly, the maintainer seems to consider these bugs to be features.
> And they've spread to other lists as well. Sigh.

One man's bug is another man's feature, I guess... but since the Reply-To:
header and most of the other things you mention are Listproc configuration
options, it's a little unfair to refer to its behavior as "broken", "bogus",
etc.

The analogy may be flawed, but I liken a mailing list to a face-to-face meeting
around a table. In a highly structured meeting, the participants may need
permission from the chairman before speaking (i.e., like a moderated list). In
a less structured meeting, anyone who wants to speak can do so at any time
(while observing some minimal rules of etiquette, of course), but they address
their remarks to everyone around the table, as on an unmoderated mailing list.
However, when a meeting degenerates into a bunch of simultaneous private
conversations, it has lost much of its value as a discussion mechanism to the
group as a whole. More often than not, if you have something to say about a
message posted to a list you're on, your comments will be of interest to many other
members of the list, assuming the original message was on-topic. If the
default behavior is for replies to go to the originator of the message, you may
forget, or not bother, to cc: the list with your message when it's of general
interest (and if you do remember, then the originator now has two copies of
your message to deal with).

My feeling has always been that the current setup of this list wrt the
Reply-To: header promotes a more open and useful discussion. The downside is
that a greater number of inappropriate (and occasionally embarrassing!)
messages get posted to the list, but such messages are easy to delete. They
wouldn't happen so often if people would get in the habit of peeking at the To:
header before hitting the Send button (yeah, I know... dream on :-). On the
other hand, you can't conjure up those potentially useful messages that you never
saw because they didn't go to the list.

I realize this is a sore point - we're all getting more junk email these days,
and we all want to minimize it. I'm not the maintainer of this list, but
perhaps he or someone else on the list could conduct a poll to see whether the
majority of subscribers want to change the Reply-To behavior of the list. This
would be done by private email, of course - votes posted to the list would NOT
be counted - but they would tend to prove the thesis that the list's behavior
needs changing. :-)

Barry

--
  Barry McLarnon VE3JF/VA3TCP | Internet: bm@hydra.carleton.ca
  Ottawa Amateur Radio Club | AMPRnet: bm@lynx.ve3jf.ampr.org
  Packet Working Group | Web: http://hydra.carleton.ca

160. Re: [HFSIG:2370] Re: Speech coding (was SQPSK)

From: Dave Emery <die@pig.die.com>
Date: Tue, 29 Jul 1997 00:55:54 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199707290455.AAA32508@pig.die.com>
Rendered body type: text/plain

Phil Karn wrote
>
> I tend to agree with the rest of your message, especially the stuff
> about the content-free nature of much of ham radio and USENET...

        Unfortunately, Greshams law applies and the bad drives out
or drowns out the good.

>
> Phil
>

161. EVM56K for sale

From: Kory Hamzeh <kory@avatar.com>
Date: Wed, 30 Jul 1997 14:33:51 -0700 (PDT)
To: dsp@tapr.org, hfsig@tapr.org
Message-ID: <Pine.BSI.3.91.970730143301.21368A-100000@avatar.avatar.com>
Rendered body type: text/plain

I have an EVM56K here which is gathering dust. It includes all of the docs
and software. This was not through the TAPR purchase. I would like to get
$125 + shipping for it. It has been socketed for the flash chip and I'll
include the flash chip.

Thanks,

Kory
AC6RN

162. Re: [HFSIG:2371] List Behavior [Was Re: SQPSK]

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 30 Jul 1997 15:03:23 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707302203.PAA06846@servo.qualcomm.com>
Rendered body type: text/plain

>One man's bug is another man's feature, I guess... but since the Reply-To:
>header and most of the other things you mention are Listproc configuration
>options, it's a little unfair to refer to its behavior as "broken", "bogus",
>etc.

Okay, then the Listproc configuration options for this list are broken and bogus, not listproc itself.

RFC-822 and RFC-1123 specifically say that mail headers should not be modified at all by mail transfer agents (this includes mailing list processors) except by the addition of a Received: header to audit the path it took.

The Reply-To: header is specifically intended for use by a sender when he wants personal replies to be sent to an address other than the one he's sending from (which appears in the From: field). True, many users accomplish the same thing without Reply-To: by simply lying about their From: address (this is easy to do in Eudora). But if a sender does put a Reply-To: header on his message, the list processor destroys it in the process of adding its own Reply-To: hfsig@tapr.or header. This makes it IMPOSSIBLE -- not just difficult -- to correctly send a private reply to the sender. This alone should be enough to ban the practice.

As for noise on the mailing lists, I find it hard to understand how anyone who uses the Internet on a regular basis could support a feature that increases the general level of traffic. Mailing list (and usenet) interfaces should go out of their way to encourage private replies to the sender. You *should* have to go a little out of your way to reply to the whole group. The idea is to make you pause for at least a fraction of a second to consider whether the whole group really needs to see your response.

By the way, this list violates the header-modification rule in other ways too -- by stripping out all other recipients from the To: and Cc: lines, and by munging the Subject line. From personal experience I can say that these practices make it *much* harder to build email tools to automatically catalog email and strip out duplicates.

Phil

163. Re: [HFSIG:2347] Re: SQPSK

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 30 Jul 1997 18:15:36 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199707310115.SAA07844@servo.qualcomm.com>
Rendered body type: text/plain

Brian Beezley asked:

>What is the channel separation theorem?

I've asked several people who ought to know (Jack Wolf and Andrew Viterbi) and surprisingly enough they haven't been able to point me to an explicit statement of this theorem in Shannon's papers. I also looked myself but couldn't find anything.

But it's implicit enough in Shannon's work, and Viterbi stated it as well as anybody could in his 1990 IEEE Communications Magazine article:

> 2.Completely separate techniques for digital source compression from those for channel transmission even though the first removes redundancy and the second inserts it.

The complete paper is on my web page:

<http://people.qualcomm.com/karn/viterbi_lessons.html>

--Phil

164. Re: [HFSIG:2374] Re: List Behavior [Was Re: SQPSK]

From: Ed Lyons <eel@ipass.net>
Date: Wed, 30 Jul 1997 21:28:57 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970731012857.00699c90@ipass.net>
Rendered body type: text/plain

Phil,

I have been trying to UNSUBSCRIBE to hfsig for over two months now, but
cannot seem to do it. I get undeliverable with a reply to change the header
of my mail address. I have tried everything, even following the addressing
instructions returned with failed messages. Can and would you help me
unsubscribe to hfsig, or point me in still another direction, or give me an
eMail address of someone who can help me.

I would SINCERELY appreciate it. Thanks very much.

Ed LYons
KD6IET, Raleigh NC.

At 05:10 PM 7/30/97 -0500, you wrote:
>>One man's bug is another man's feature, I guess... but since the Reply-To:
>>header and most of the other things you mention are Listproc configuration
>>options, it's a little unfair to refer to its behavior as "broken", "bogus",
>>etc.
>
>Okay, then the Listproc configuration options for this list are broken
>and bogus, not listproc itself.
>
>RFC-822 and RFC-1123 specifically say that mail headers should not be
>modified at all by mail transfer agents (this includes mailing list
>processors) except by the addition of a Received: header to audit the
>path it took.
>
>The Reply-To: header is specifically intended for use by a sender when
>he wants personal replies to be sent to an address other than the one
>he's sending from (which appears in the From: field). True, many users
>accomplish the same thing without Reply-To: by simply lying about
>their From: address (this is easy to do in Eudora). But if a sender
>does put a Reply-To: header on his message, the list processor
>destroys it in the process of adding its own Reply-To: hfsig@tapr.or
>header. This makes it IMPOSSIBLE -- not just difficult -- to correctly
>send a private reply to the sender. This alone should be enough to ban
>the practice.
>
>As for noise on the mailing lists, I find it hard to understand how
>anyone who uses the Internet on a regular basis could support a
>feature that increases the general level of traffic. Mailing list (and
>usenet) interfaces should go out of their way to encourage private
>replies to the sender. You *should* have to go a little out of your
>way to reply to the whole group. The idea is to make you pause for at
>least a fraction of a second to consider whether the whole group
>really needs to see your response.
>
>By the way, this list violates the header-modification rule in other
>ways too -- by stripping out all other recipients from the To: and Cc:
>lines, and by munging the Subject line. From personal experience I can
>say that these practices make it *much* harder to build email tools to
>automatically catalog email and strip out duplicates.
>
>Phil
>
>
>

165. Unsubscribe

From: "Norman R. Johnson" <normj@interaccess.com>
Date: Wed, 30 Jul 1997 22:06:08 -0500
To: hfsig@tapr.org
Message-ID: <3.0.3.32.19970730220608.006a7120@POP3.interaccess.com>
Rendered body type: text/plain

Please help,

I have been trying to unsubscribe for the last 4 weeks.

Please help.

May the Good Lord Bless You

Rick

166. Re: [HFSIG:2374] Re: List Behavior [Was Re: SQPSK]

From: Barry McLarnon VE3JF <bm@lynx.ve3jf.ampr.org>
Date: Thu, 31 Jul 1997 03:51:36 +0000 (GMT)
To: hfsig@tapr.org
Message-ID: <199707310351.DAA18598@lynx.ve3jf.ampr.org>
Rendered body type: text/plain

Phil Karn <karn@qualcomm.com> wrote:
> Okay, then the Listproc configuration options for this list are broken
> and bogus, not listproc itself.
>
> RFC-822 and RFC-1123 specifically say that mail headers should not be
> modified at all by mail transfer agents (this includes mailing list
> processors) except by the addition of a Received: header to audit the
> path it took.

I disagree with your contention that listservers such as listproc and majordomo
are MTAs. If a mailing list is implemented by means of an MTA's alias mechanism,
then it's true that only the MTA is involved in processing the mail. Listproc
and its ilk are mail-based applications, not MTAs, so all bets are off regarding
the relationship between their inputs and outputs.

> The Reply-To: header is specifically intended for use by a sender when
> he wants personal replies to be sent to an address other than the one
> he's sending from (which appears in the From: field). True, many users
> accomplish the same thing without Reply-To: by simply lying about
> their From: address (this is easy to do in Eudora). But if a sender
> does put a Reply-To: header on his message, the list processor
> destroys it in the process of adding its own Reply-To: hfsig@tapr.or
> header. This makes it IMPOSSIBLE -- not just difficult -- to correctly
> send a private reply to the sender. This alone should be enough to ban
> the practice.

I agree this is a weakness, but only a minority of list participants will
have Reply-To: headers on their messages, and they will typically include
the address to reply to in their signature. Yes, it's an inconvenience if
you want to send them a private reply, but I still firmly believe that the
majority of replies should go to the whole list (assuming it's a high-SNR
list). Why would you be sending someone an irrelevant reply?

> As for noise on the mailing lists, I find it hard to understand how
> anyone who uses the Internet on a regular basis could support a
> feature that increases the general level of traffic. Mailing list (and
> usenet) interfaces should go out of their way to encourage private
> replies to the sender. You *should* have to go a little out of your
> way to reply to the whole group. The idea is to make you pause for at
> least a fraction of a second to consider whether the whole group
> really needs to see your response.

Well, that pause should be second nature, before you send *any* email.
I still think that the default destination for replies should be the list,
in most cases - perhaps not for high-volume, noisy lists though.

> By the way, this list violates the header-modification rule in other
> ways too -- by stripping out all other recipients from the To: and Cc:
> lines, and by munging the Subject line. From personal experience I can
> say that these practices make it *much* harder to build email tools to
> automatically catalog email and strip out duplicates.

Again, the listservers don't have to follow the rules that an MTA does - they
could receive email and generate a response in smoke signals. This might
actually be an improvement for some lists. :-) I don't see much problem
with filtering list mail and sorting it into folders and such - there are
enough headers available for that.

This isn't a religious issue with me - about half the lists I subscribe to are
set up the way this one is, and the other half are set up the way Phil prefers.
I'll happily go along with whatever the majority of this list's users want.
A quick poll conducted off-list is probably the best way to find out.

Barry

--
  Barry McLarnon VE3JF/VA3TCP | Internet: bm@hydra.carleton.ca

```
  Ottawa Amateur Radio Club      |  AMPRnet:  bm@lynx.ve3jf.ampr.org
  Packet Working Group           |  Web:      http://hydra.carleton.ca
```

167. Re: [HFSIG:2376] Re: List Behavior [Was Re: SQPSK]

```
From: forrerj@peak.org (Johan Forrer)
Date: Wed, 30 Jul 1997 22:18:56 -0800
To: hfsig@tapr.org
Message-ID: <199707310518.WAA08438@peak.org>
Rendered body type: text/plain
```

Ed,

Two things you can try that may do the trick:

1) You have to unsubscribe with the same address that you subscribed with originally. Otherwise one person can unsubscribe another and that won't work.

2) Have you tried going to TAPR's web page (www.tapr.org) with your web browser, go to Special Interest Groups section. There is a box you can request to be removed from the any list (provided you are signed on with the same e-mail address that you subscribed with of course).

3) Let us know if this does not work, or you cannot sign on as your original address, we'll try do something else.

Thanks,

--Johan Forrer

>Phil,
>
>I have been trying to UNSUBSCRIBE to hfsig for over two months now, but
>cannot seem to do it. I get undeliverable with a reply to change the header
>of my mail address. I have tried everything, even following the addressing
>instructions returned with failed messages. Can and would you help me
>unsubscribe to hfsig, or point me in still another direction, or give me an
>eMail address of someone who can help me.
>
>I would SINCERELY appreciate it. Thanks very much.
>
>Ed LYons
>KD6IET, Raleigh NC.
>

168. Re: [HFSIG:2379] Re: List Behavior [Was Re: SQPSK]

```
From: Hasch@t-online.de (Juergen Hasch)
Date: Thu, 31 Jul 1997 12:08:16 +0200
To: hfsig@tapr.org
Message-ID: <33E06410.FCC27AED@t-online.de>
Rendered body type: text/plain
```

Johan Forrer wrote:

> Ed,
>
> Two things you can try that may do the trick:
>
> 1) You have to unsubscribe with the same address that you subscribed
> with
> originally. Otherwise one person can unsubscribe another and that
> won't work.

Hello Johan,I tried to unsubscribe from my university mail account a
while ago, and it just
wouldn't work.
The problem is that sometimes only your email receive address is common,
but
when you are sending emails, your address may depend on the computer you

are working on.
I finally unsubscribed by telnetting to the smtp mailserver at tapr.org
and typing in
my email address :-)

73 de Juergen   DG1SCR

169. Re: [HFSIG:2375] Re: SQPSK

From: Antony Martin <anthony.martin@gecm.com>
Date: Thu, 31 Jul 1997 13:50:49 +0100
To: hfsig@tapr.org
Message-ID: <970731135049.18708@oas01.gmcl.co.uk>
Rendered body type: text/plain

>
> 2.Completely separate techniques for digital source compression from
> those for channel transmission even though the first removes
> redundancy and the second inserts it.
>
> --Phil
>

This is fine as a theory. I like it very much.

But just like the shannon limit, it's an unattainable goal.

In a theoretical universe, both compression & correction
would be perfect and so the rule would work fine. But they're
not, and performance may be improved if we ignore the rule.

1) No error correction scheme reduces errors to zero. Only
infinite SNR or zero data rate reduces errors to zero. (Shannon's
theory). Hence the compression scheme must be able to tolerate
the remaining errors without severe loss of the data.

2) No compression scheme can reduce redundancy to zero. To
remove all duplication the compressor must have access to all
data for the link for all time which implies knowledge of the
future, which violates the causality rule. The requirement for
a finite delay means we must accept much less compression.

3) Compressed data doesn't have bits that are equally significant.
Again, we're limited by finite algorithms.

5) Parameters of the compression and transmission combine to
determine the overall characteristics of the link. The two can't
be designed independently.

4) Comms schemes may be one-to-many or many-to-one or having
multiple routes, in which case redundancy may exist as a
function of the topology, rather than part of the coding scheme.

6) Caches are a means of reducing redundant traffic at the
receiving end, not at the source. Is this not 'compression'?
Why should compression have to be at the source?

Anthony.

Having said this, it's a bit academic as many amateur modes have
far more basic problems. eg. variable length codes, ability to
obtain inverted data or false sync, symbols that are not equiprobable...

170. Re: [HFSIG:2380] Re: List Behavior [Was Re: SQPSK]

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Thu, 31 Jul 1997 09:00:57 -0600 (MDT)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.3.96.970731085802.26123A-100000@murdrum>
Rendered body type: text/plain

> The problem is that sometimes only your email receive address is common,
> but
> when you are sending emails, your address may depend on the computer you
>
> are working on.

Yes, this is a common problem for me as well. My email address highly
depends on where I send email from. In fact, many people insist that my
email address at work is tim@aa5df.sps.mot.com despite my signature files
and reply-to lines. This is not correct since aa5df just happens to be the
name of the workstation, but I may want to send email from any number of
systems here.

> I finally unsubscribed by telnetting to the smtp mailserver at tapr.org

> and typing in
> my email address :-)

Done the mail spoof thing myself a few times... Sometimes you just need a
bigger sledge hammer!

73
Tim

tim@acca.nmsu.edu

August 1997

Original source file: hfsig.9708
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9708

1. FOR LYLE JOHNSON

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Fri, 01 Aug 1997 09:47:57 -0400
To: hfsig@tapr.org
Message-ID: <33E1E90D.5FB04ECC@worldnet.att.net>
Rendered body type: text/plain

Lyle:

Mail to your azstarnet.com has bounced from both myself and two others
who
were trying to reply to email.

Bob

--
****

* Robert W. McGwier, Ph.D. * Mathematician. Hobbies: Computers,
Astronomy*
* 64 Brooktree Rd. * Amateur Radio, and Golf. Scouts: Advisor
*
* East Windsor, NJ 08520 * Post 995, Troop 5700 Advancement Chair,
*
* (H) 609-443-8963 * Sanhican #2 (Brotherhood), Used to be a
*
* (F) 609-924-3061 * Buffalo (NE-III-120). George Washington
*
* (W) 609-279-6240 * Council. Brownie Troop 193
*
****

2. query

From: Kaprocki <fkaprocki@iag.net>
Date: Mon, 04 Aug 1997 17:17:36 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970804211736.006659ac@iag.net>
Rendered body type: text/plain

I s hfsig the same as hfsig-tnc? Are these two distinct groups? Where can I
get a list of the groups? tnks,73's kk4rw

3. subscribe

From: Kaprocki <fkaprocki@iag.net>
Date: Mon, 04 Aug 1997 17:15:12 -0400
To: hfsig@tapr.org
Message-ID: <1.5.4.32.19970804211512.0067f404@iag.net>
Rendered body type: text/plain

[Empty message body]

4. Out of topic: Tektronix 2213 Service Manual

From: Eduardo Jacob <edu@kender.es>
Date: Wed, 06 Aug 1997 22:27:13 -0100
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19970806222713.006ad488@kender.es>
Rendered body type: text/plain

This is out of topic, but as the audience is rather selected perhaps I
could get help here.
I have been lend an old Tektronix 2213A 60Mhz osciloscope and it is a
little unadjusted. Does anybody have an idea of where to get a service
manual of it?, I haven't asked directly tektronix neither to calibrate nor
to get the manual from them as the price in both cases migth be rather high.

Eduardo/EA2BAJ

5. Re: [HFSIG:2381] Re: SQPSK

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 6 Aug 1997 17:55:45 -0700 (PDT)
To: hfsig@tapr.org

Cc: karn@qualcomm.com
Message-ID: <199708070055.RAA06462@servo.qualcomm.com>
Rendered body type: text/plain

>In a theoretical universe, both compression & correction
>would be perfect and so the rule would work fine. But they're
>not, and performance may be improved if we ignore the rule.

Please cite some examples. Real systems can and do very well by
separating source and channel coding. It's not only the right thing
from an information theory perspective, it's also the right thing from
a computer science perspective as it promotes modularity and clean
protocol layering. Some successful real-world examples include CDMA
digital cellular and direct digital satellite broadcasting. The latter
is a direct descendent of image transmission techniques originally
used in deep space exploration, particularly Voyager.

>1) No error correction scheme reduces errors to zero. Only
>infinite SNR or zero data rate reduces errors to zero. (Shannon's
>theory). Hence the compression scheme must be able to tolerate
>the remaining errors without severe loss of the data.

This is somewhat misleading. Shannon showed that you can make the
error rate *arbitrarily small* on a noisy channel as long as you
signal at less than the channel capacity. Real, easily-implemented
codes can make "arbitrarily small" quite small indeed. Many charts of
Reed-Solomon code performance show uncorrected BER values on the order
of 10^-20 or less when the blocks are large enough.

If error detection (as opposed to correction) is sufficient, you can
make do with a CRC having much less overhead and delay. Our experience
in CDMA has been that even a weak CRC (e.g., we use a 12-bit CRC in
our full-rate 9.6kb/s frame) is plenty in combination with other
methods (modem SNR estimation, Viterbi decoder metric thresholds) to
make undetected errors *very* rare events. "Erased" frames (frames
discarded because of CRC or other errors) are much more common, typically
1-2%, but these are tolerable because the vocoder can "ride through"
these as long as they are in fact flagged as erasures.

>2) No compression scheme can reduce redundancy to zero. To
>remove all duplication the compressor must have access to all
>data for the link for all time which implies knowledge of the
>future, which violates the causality rule. The requirement for
>a finite delay means we must accept much less compression.

This is much more true than #1. It's the reason our vocoder *can*
tolerate our nominal 1-2% frame erasure rates.

>3) Compressed data doesn't have bits that are equally significant.
>Again, we're limited by finite algorithms.

Again, true. E.g., an error in gzipped data generally means that everything
past the error is trashed. So if you're going to have an error in gzipped
data, you want it to be as near the end as possible so you get as much
uncorrupted data as possible.

>5) Parameters of the compression and transmission combine to
>determine the overall characteristics of the link. The two can't
>be designed independently.

Mostly true, if you consider the FEC and ARQ mechanisms (if any) to be
part of the transmission link. In general, the better the FEC the
longer the data blocks it will want to work with, and it will either
decode these blocks perfectly or not at all. So you can still attain a
certain degree of independence by establishing a packet-like interface
between the compression and transmission layers.

>4) Comms schemes may be one-to-many or many-to-one or having
>multiple routes, in which case redundancy may exist as a
>function of the topology, rather than part of the coding scheme.

True, though this could be seen as simply a special form of repetition
coding -- which is known to be much less efficient than less constrained
coding schemes (where the redundancy is a more complex function of the
original data.)

>6) Caches are a means of reducing redundant traffic at the
>receiving end, not at the source. Is this not 'compression'?
>Why should compression have to be at the source?

Caching is just a special case of a more general compression
mechanism, which inherently requires state at both the sender and the
receiver -- the greater the compression, the more state is probably
required at both ends. And the more state you have at the two ends,
the more vital it is to keep it properly synchronized -- meaning the
less tolerant you are of transmission errors.

>Having said this, it's a bit academic as many amateur modes have
>far more basic problems. eg. variable length codes, ability to
>obtain inverted data or false sync, symbols that are not equiprobable...

Agreed. But even academic discussions have their place in keeping you
focused on the right long-term goals.

Back to the separation theorem -- the only reason I'm aware of for not
exploiting it as much as you possibly can is the delay that's
inherently involved. E.g., IS-95 CDMA pays a price in delay for its
performance. Each vocoder frame is 20ms long, and by the time you add
the inevitable implementation delays (store-and-forward delays within
the system, Viterbi decoding delays, vocoder execution time, etc)
you're pushing something like 100ms, which some people start to
notice. Nevertheless, this delay still buys a substantial amount of
performance.

Much greater delays can be found in high performance imaging systems
like DSS. By watching the same channel on my local cable system and on
DSS I've measured the extra DSS delay (over and above the extra
satellite hop) at about 1 second, but this is not a problem in a one-way
broadcast situation.

Phil

6. Re: [HFSIG:2386] Out of topic: Tektronix 2213 Service Manual

From: Jim Pruitt <amburch@owt.com>
Date: Wed, 6 Aug 1997 23:08:59 -0700
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19970806230215.08a72a00@oneworld.owt.com>
Rendered body type: text/plain

Try Deane Kidd, W7TYR
    27270 S.W. Ladd Hill Rd.
    Sherwood, Or 97140
    (503) 625-7363

        Deane is retired from Tektronix and has a business of supplying
manuals for older Tek equipment. I do not know what he would charge but you
might contact him.
        Good luck.
        Jim Pruitt, WA7DUY

At 04:40 PM 8/6/97 -0500, you wrote:
>This is out of topic, but as the audience is rather selected perhaps I
>could get help here.
>I have been lend an old Tektronix 2213A 60Mhz osciloscope and it is a
>little unadjusted. Does anybody have an idea of where to get a service
>manual of it?, I haven't asked directly tektronix neither to calibrate nor
>to get the manual from them as the price in both cases migth be rather high.
>
>   Eduardo/EA2BAJ
>
>

7. TAPR Server to be down 8/9/97

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Fri, 8 Aug 1997 22:24:44 -0500
To: "TAPR-BB list   mailing"<tapr-bb@tapr.org>,
        "TAPR   APRS SIG"<aprssig@tapr.org>,
        "HF SIG list   mailing"<hfsig@tapr.org>,
        "NETSIG list   mailing"<netsig@tapr.org>,
        "BBS SIG list   mailing"<bbssig@tapr.org>,
        " Spread Spectrum "<ss@tapr.org>, " TAPR SYS "<tapr-sys@tapr.org>,
        " tapr-tnc "<tapr-tnc@tapr.org>, aprs-fl@tapr.org,
        " tacgps "<tacgps@tapr.org>,
        TAPR Regional Freq <regional_freq@tapr.org>,
        " TAPR/AMSAT DSP "<dsp@tapr.org>, " DSP-93 Build "<dsp-93@tapr.org>,
        " TPRS Board "<tprs-board@tapr.org>
Message-ID: <v03020900b01191cb3c5c@[208.134.134.40]>

Rendered body type: text/plain

I just received a call informing us that the building location where the TAPR.ORG server is located will be without power tomorrow for most of the day.

We will be shutting down the server between 7am and 8am and don't expect that it will be available until late Saturday evening when power is restored.

Cheers - Greg, WD5IVD

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd

"Before you become too entranced with gorgeous gadgets
and mesmerizing video displays, let me remind you that
information is not knowledge, knowledge is not wisdom,
and wisdom is not foresight. Each grows out of the other
and we need them all."   -Arthur C. Clarke

8. Static Crashes and FEC

From: Charles Brain <chbrain@dircon.co.uk>
Date: Sun, 10 Aug 1997 10:42:24 +0100
To: hfsig@tapr.org
Message-ID: <33ED8D00.669F5890@dircon.co.uk>
Rendered body type: text/plain

Hello,

I know this is not directly about HF but here goes.

I have just started doing some experiments on the U.K 73 Khz band. So far I have managed to receive aurol CW from an amateur station about 50 miles away.

I am now looking at data transmission in this band. I intend using BPSK and convolutional encoding. The main limiting factor seems to be that static crashes on this band are 50 db+ stronger than the wanted signal. There is also TV timebase/switched mode power supply interference as well, but this only starts around lunchtime.

I will be down converting the 73 Khz signal into the audio range using a complex mixer based on a couple of MC1496 mixer chips. The local oscillator will use a DDS chip fed into a couple of D type flip-flops to get the sin(wt) and cos(wt) signals.
The I and Q output is then being fed into the stereo channels of a Motorola EVM. The error correction will be done on a P.C.
I am hoping to make a VLF/LF transceiver that operates from about 3 Khz to 200 Khz as a by-product.

What I am proposing to do is to detect the static crashes and feeding them as erasures to the Viterbi decoder.

The question is, should I be looking a a high data rate (75b/s) and lots of FEC or a lower data rate (5b/s) and less FEC. There is little or no multipath on the band so that isn't a problem and the bulk of the communication is using surface wave. The Q of the trasmit antenna restricts the bandwidth to about 300 Hz maximum. It is also relatively easy to find clear frequencies on the band. Again most of the interference is local (noise cancelling techniques may prove useful here).

I think probably that using the higher data rate is the way to go as with the lower data rate the static crash Impulses cause the filters to ring.

Another question is what is the best way to get soft decision information from the demodulator (Costas Loop). In the example in my text book it uses a channel where 1 is -1v and 0 is +1v it is therefore simple to calculate the certainties. However I will get a positive or negative voltage coming out of the demodulator, how do I convert this into a probability as I have no fixed level. All I know (maybe) is that the greater the magnitude the more certain the result. Or should I just use 1,0,Erasure and leave it at that.

I also read that when trying to detect weak signals in the presence
of very strong signals it is better to use a logarithmic response
rather than AGC. Any comments? I was hoping to get away with
no AGC for demodulating data. I will be digitising the entire band
which includes a military MSK transmitter right in the center. I am
hoping the dynamic range of the codec will be sufficient.

The above questions touch on a number of things that have been
discussed on the sig so I hope it is not considerd too weird!

- Charles G4GUO

9. unscribe

From: Manuel Fernando Resende Pereira <nop04918@mail.telepac.pt>
Date: Thu, 07 Aug 1997 11:32:31 +0100
To: hfsig@tapr.org
Message-ID: <33E9962F.28FB@mail.telepac.pt>
Rendered body type: text/plain

Best regards.
Please unscribe.

10. 1997 DCC Call for Papers (Reminder)

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Mon, 11 Aug 1997 00:00:32 -0500
To: "TAPR-BB list   mailing"<tapr-bb@tapr.org>,
        "HF SIG list   mailing"<hfsig@tapr.org>,
        " Spread Spectrum "<ss@tapr.org>
Message-ID: <v03110704b0144c672205@[208.134.134.40]>
Rendered body type: text/plain

Call for Papers

1997 ARRL and TAPR Digital Communications Conference
October 10-12, 1997
Baltimore, Maryland (minutes from BWI airport)

Web: http://www.tapr.org/dcc

It's that time again! Time to mark your calendar and think about what to
publish for the upcoming 16th Annual ARRL and TAPR Digital Communications
Conference.

The 1997 ARRL and TAPR Digital Communications Conference will be held
October 10-12, 1997 in Baltimore, Maryland. This year's conference location
is just minutes away from the BWI (Baltimore/Washington International)
Airport.

For full details on the conference see http://www.tapr.org/dcc or contact
the TAPR office for the DCC information flyer.

Call for Conference Proceeding Papers

Anyone interested in digital communications is invited to submit a paper
for publication in the Conference Proceedings. Presentation at the
Conference is not required for publication. If you know of someone who is
doing great things with digital communications, be sure to personally tell
them about this! Papers are due by August 20th, 1997, and should be
submitted to Maty Weinberg, ARRL, 225 Main Street, Newington, CT 06111 or
via the Internet to lweinberg@arrl.org. Information on paper submission
guidelines are available on-line (http://www.tapr.org/dcc). Submissions
can be e-mailed.

Contact TAPR to register of for more information on the DCC.

Tucson Amateur Packet Radio, 8987-309 E. Tanque Verde Road #337, Tucson, AZ
85749-9399. Phone: (940) 383-0000. Fax: (940) 566-2544.
Internet: tapr@tapr.org   http://www.tapr.org

Tucson Amateur Packet Radio

8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399 * 940-383-0000
---
e-mail: tapr@tapr.org ftp: ftp.tapr.org web: <http://www.tapr.org/>
---

11. Progress on HF BPSK

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 11 Aug 1997 10:36:04 -0800
To: HFSIG@tapr.org, DSP@tapr.org
Message-ID: <199708111736.KAA19433@peak.org>
Rendered body type: text/plain

Hi folks,

Thought that I would let you know what's been happening with HF BPSK experimentation.

1) Dave, AA1A and Peter, G3IRM, made a transatlantic contact using BPSK (40 baud, synchronous ASCII.)
Congratulations!

2) Bill, WB6RIJ, is now also up and running with an EVM. We have had several QSO's on 30m and 40m under good and also very poor conditions. Next we would like to experiment with the European PSK31 system, also Pawel's new MT63ASC (63 tone BPSK) modem. I'll let you know how that works out.

3) After receiving reports about excessive spectrum usage, the COHERENT BPSK group has actively been working on refining modulation techniques to reduce bandwidth to a minimum.

The basic problem apparently arise from the use of square symbol keying, in addition to inadequate waveform filtering. The present solution that is being worked on is based on using a PC sound card to generate shaped symbols. Tests have shown a great deal of improvement and eventually should become quite acceptible to others on the band segment - that is once everyone does what is needed to clean up their emissions.

Folks that use the EVM solution has several options to choose from: narrow FIR bandpass filtering or shaped symbols. Either will work, however, there is no free lunch here - several side effects to contend with: a) the demodulator needs to use a matched filter, b) amplitude variations are becoming a factor - as a matter of fact, one cannot have both narrow bandwidth and constant amplitude simultaneously, c) linearity in the amplifier chain becomes a factor, d) frequency stability becomes even more of an issue.

It is evident that weak signal performance deteriorates rather quickly when these factors are overlooked.

4) I will shortly be updating the BPSK interface code with COHEVM12.ZIP. This has several improvements over the older version. I'll make the announcement when its ready.

5) The TAPR EVM radio interface kit is making good progress. Hopefully this will enable those that have been putting off the construction of their own interface to become active.

For those that are interested, most BPSK activity is presently on 7081 (carrier) in the evenings. Presently there is some activity on 14081 (carrier) at 1600Z to work on transatlantic contacts. There are stations around the North America; Quebec, Massacusetts, Colorado, Oregon, and California.

73

--Johan

12. RE: [HFSIG:2393] Progress on HF BPSK

From: jalocha@axdele.cern.ch
Date: Mon, 11 Aug 1997 22:23:27 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199708112023.WAA22695@dxmint.cern.ch>
Rendered body type: text/plain

Actually, Paul, PA0OCD and Izak, PA0IHD has made a QSO with my 64-tone, 10 baud modem. It was on 80 m on a distance of about 100 km. With Paul's permition I put here the relevant piece of his report.

Pawel

p.s. Paul e-mail is paulstra@xs4all.nl

==========

Hello Pawel, PA0IHD and me had this evening our first 64 tones bpsk qso !.
I was working in the end wit abt 2 watts !. and that on 80 mtrs with an
indoor magnetic loop (which is actually not ment for 80 hi, is only 1,5 by
1,5 meters)
But perhaps the poweroutput is an bit difficult to see with 64 tones.

O yes to get on qrg we used first slowbpsk psk31 so we get witin 1 hz on
qrg.

Our first impression:

- Very very good , with much qrm with amtor arq , cw etc.
(I think better then psk31 , but that has nor error correction at all)

- The drift: no problem holding lock, but it woukd be nice to have an
control-function so we can see the difference in qrg during qso, and/or
have an function key to start lock again.
I saw after 1 hour (!) an qrg difference of 30 hz, so first I made Signal
lost and made an qrg adjustment and locked again, did it 2 times and was
within 1 hz again hi.

_ Tuning aid would be nice, perhpas an special tune mode ? with 1 tone ?

[...]

73 de Paul, pa0ocd

==========

13. Re: [HFSIG:2390] Static Crashes and FEC

From: Antony Martin <anthony.martin@gecm.com>
Date: Mon, 11 Aug 1997 22:11:55 +0100
To: hfsig@tapr.org
Message-ID: <970811221155.10087@oas01.gmcl.co.uk>
Rendered body type: text/plain

> Hello,
>
> I know this is not directly about HF but here goes.
>
> I have just started doing some experiments on the U.K 73 Khz
> band. So far I have managed to receive aurol CW from an amateur
> station about 50 miles away.
>
> I am now looking at data transmission in this band. I intend using
> BPSK and convolutional encoding. The main limiting factor seems
> to be that static crashes on this band are 50 db+ stronger than the
> wanted signal. There is also TV timebase/switched mode power
> supply interference as well, but this only starts around lunchtime.
>
> I will be down converting the 73 Khz signal into the audio range using a
> complex mixer based on a couple of MC1496 mixer chips. The local
> oscillator will use a DDS chip fed into a couple of D type flip-flops
> to get the sin(wt) and cos(wt) signals.
> The I and Q output is then being fed into the stereo channels of a
> Motorola EVM. The error correction will be done on a P.C.
> I am hoping to make a VLF/LF transceiver that operates from about
> 3 Khz to 200 Khz as a by-product.

Remember your mixers won't be exactly matched. If you attempt SSB
demodulation your opposite sideband rejection is likely to be only
around 30dB. Hence design your modulation to use both sides of the
carrier (ie. ISB). Don't rely on zero (ie. carrier) frequency as
the dc will shift about all over the place. Your EVM probably can't
convert DC in anycase - how low frequency does it go?

Does the EVM sample both channels simultaneously? If not you have an
unnecessary awkward problem.

74HC4043 analogue switches are doing my I/Q mixing at 500kHz just fine...

> I also read that when trying to detect weak signals in the presence
> of very strong signals it is better to use a logarithmic response

> rather than AGC. Any comments? I was hoping to get away with
> no AGC for demodulating data. I will be digitising the entire band
> which includes a military MSK transmitter right in the center. I am
> hoping the dynamic range of the codec will be sufficient.

You've hit the nail on the head--- this is the basic problem and goal of
all receiver design. The crux of the matter. The difference between a
pocket tranny and a comms receiver. Performance comes from
knowing when to mix, when to filter, when to amplify and when to
digitise. And what to do to get the LOs clean enough.

DDS synthesizers normally need a clean-up PLL; an ADC with 70dB SFDR
must still appear after the channel filter. (Maybe I'm being too severe
- - that's my military/professional radio background.)

Despite all this wonderchips digital stuff, you still get best performance
with a fractional-N PLL synth, ananalogue channel filter, and AGC. If you
digitise too early you won't get the dynamic range as the linearity of
the converter won't be high enough.

If your receiver antenna has a narrow BW like you say then much of the
out-of-band stuff will get attenuated.

must go, Anthony.

14. Re: [HFSIG:2392] 1997 DCC Call for Papers (Reminder)

From: w2pat@juno.com
Date: Mon, 11 Aug 1997 09:18:52 -0400
To: hfsig@tapr.org
Message-ID: <19970811.091854.7542.2.w2pat@juno.com>
Rendered body type: text/plain

[Empty message body]

15. Re: [HFSIG:2395] Re: Static Crashes and FEC

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 11 Aug 1997 16:49:11 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199708112349.QAA13690@servo.qualcomm.com>
Rendered body type: text/plain

> rather than AGC. Any comments? I was hoping to get away with
> no AGC for demodulating data. I will be digitising the entire band

AGCs are a problem. You particularly don't want static crashes
knocking the gain way down for a while so you lose more channel
symbols even after the crash ends. Better to let the crash saturate
the receiver, as long as it recovers quickly afterwards.

But you do want the desired signal to be within a reasonable range.
So I suggest a smart AGC driven by the demodulator's estimate of the
desired signal, not by raw signal energy. Also, if you use soft
decision decoding you want a way to detect static crashes and force
the resulting demodulated symbols to erasures; otherwise they could
appear to the decoder as highly confident 1s and 0s (because of their
high amplitudes) when in fact they're garbage. The same static crash
detector could help clamp the AGC.

Phil

16. Re: [HFSIG:2390] Static Crashes and FEC

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Mon, 11 Aug 1997 22:17:06 -0400
To: hfsig@tapr.org
Message-ID: <199708120217.WAA09406@aus-d.mp.campus.mci.net>
Rendered body type: text/plain

>I know this is not directly about HF but here goes.

I bet if we had a 73Khz band available in the U.S. for amateur radio we'd be
experimenting with it too. I guess the best we've got is the 160 Kz - ???,
limited to 1 watt and an antenna system length of 15 meters.

...
>I am now looking at data transmission in this band. I intend using
>BPSK and convolutional encoding. The main limiting factor seems
>to be that static crashes on this band are 50 db+ stronger than the

>wanted signal.
...
>- Charles G4GUO

I remember reading "somewhere & sometime long ago" about a scheme to deal
with impluse/crashes with high-Q, low freq receiving circuits. It was along
the lines of detecting the noise impluse via a broadband receiver much like
the usual arrangement for noise-blankers, but since the antenna tuned
circuit is a very high Q the detected noise signal was used to switch in
some damping into the antenna circuit during the impluse duration...must
have read it 30-40 years ago. The problem with the usual noise blanking is
that it is usually is placed before the last, narrow if filter, but in the
low freq stuff the narrow filter is the antenna matching itself.

(And now that I think about the issue, I wonder how the digital equivalent
of the noise-blanker might be implemented in DSP? Maybe it is a case that
with DSP better approaches are feasible, i.e. the correlation type of scheme.)

Don, W4DH

17. Re: [HFSIG:2390] Static Crashes and FEC

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 12 Aug 1997 10:31:05 +0100
To: hfsig@tapr.org
Message-ID: <33F02D59.F8380129@dircon.co.uk>
Rendered body type: text/plain

Thanks everyone for your replies,

I have been looking at the static crashes on a storage scope
they are easily distinguishable as they are thousands of times
larger than the wanted signals on the band. They seem to occur
as either single events or in groups of 4 or 5. They are
characterised as sharp pulses with an exponential decay. Most
are gone within 300us.

I was thinking of sampling the signal after the preamp using a
comparator then clamping the output when they reach a preset
level, whether I have to add a monostable after the comparator
I am not sure yet. It maybe that a Schmitt trigger will be adequate

If I clamp the signal will this be adequate for the soft decision
decoder or will I have to feed erasure information directly into
the DSP using the parallel port (taking into account the delays
through the FIR filters)? I assume the clamped signal will cause
Costas loop output to indicate a 'weaker symbol'. From Phils
comment I think feeding the blanker output into the parallel port
is the proper way to do it.

I have just got my hard decision Viterbi decoder to work with
soft decisions. It is no threat to Phils code as it is very slow,
the only real advantages it has it that the convolutional code
can be passed when it is initialised. It should be able to
accomodate up to a K=32 code. I am using the excellant DJGPP
compiler tools which generate protected mode code.

As far as DSP noise blankers are concerned, Frerking has nothing
on them. I seem to remember Antony that the Marconi DSP HF
receiver doesn't have a noise blanker either?

My original idea was to use a Xtal Oscillator to down convert to
a '10 Khz IF', filter it then use a complex mixer within the DSP code to
move the signal to the wanted frequency 'a sort of DDS in S/W'. That
however makes the DSP code a lot more complex as I have to pass
information from the host PC as to the receive frequency etc. It might
be a better way of doing it though, even though I would have to sample
at a much higher rate than at present, but using decimating filters after the
frequency conversion I can get back to something slower.

If I sample fast enough then the time offset of the A/D should be
insignificant as far as phase is concerned? I will check the Crystal
Semiconductors CS4215 data sheet. I have a PDF copy from their WWW
site. It is a 16 bit device and has a programmable gain input stage before
the A/D which has a 20 db range in 1.5 db steps.

My current fixed IF is 600Hz by the way, not DC. I am using 9K6 s/s.
Use of those analogue switches is an option I will think about.
The overall circuit would probably be much simpler and there is no need
for balancing the 1496 mixers using trimmers.

Finally Donald, yes we are allowed to run 1W EIRP on the band, a
friend of mine is working on a 1KW Class D amplifier for the band!
Using power mosfets and switched mode PSU components.
His aerial has about -48db gain! I will be using something a bit more
modest (about 100W) into a Marconi T or possibly a loop ant.

There is an annual prize for the most innovative piece of equipment
on the band Sponsered by the RSGB and Datong Electronics, but I
wouldn't be after that would I ....? After all I already qualify
for a Listeners award having heard someone more than 8 km away.
In fact I can claim a 32 Km endorsement to the award as well!

- Charles G4GUO

18. Re: [HFSIG:2399] Re: Static Crashes and FEC

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 12 Aug 1997 11:40:21 +0100
To: hfsig@tapr.org
Message-ID: <33F03D95.43052D36@dircon.co.uk>
Rendered body type: text/plain

Charles Brain wrote:

> can be passed when it is initialised. It should be able to
> accomodate up to a K=32 code. I am using the excellant DJGPP
> compiler tools which generate protected mode code.
>

Ignore the above bit about K=32 code. I don't have enough gigabytes
in my P.C to store the path memories and state information!!! I have
just done the calculation.

- Charles

19. Re: [HFSIG:2400] Re: Static Crashes and FEC

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.th-darmstadt.de>
Date: Tue, 12 Aug 1997 13:14:45 +0200
To: hfsig@tapr.org
Message-ID: <33F045A5.6D36ADBD@zeus.uet.e-technik.th-darmstadt.de>
Rendered body type: text/plain

Charles Brain wrote:
>
> Charles Brain wrote:
>
> > can be passed when it is initialised. It should be able to
> > accomodate up to a K=32 code. I am using the excellant DJGPP
> > compiler tools which generate protected mode code.
> >
>
> Ignore the above bit about K=32 code. I don't have enough gigabytes
> in my P.C to store the path memories and state information!!! I have
> just done the calculation.
>
> - Charles

I think you will have to do sequential decoding (i.e. Fano) for such a
code. Using the Viterbi algorithm will need lots of memory and time.

73' Alexander

--
------------------------------+---------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| technik u. Elektroakustik | Fax : +49-6151-165545
|
| Merckstrasse 25 | EMail: a.kurpiers@uet.th-darmstadt.de
|
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
------------------------------+---------------------------------------

20. Re: [HFSIG:2401] Re: Static Crashes and FEC

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 12 Aug 1997 14:23:28 +0100
To: hfsig@tapr.org
Message-ID: <33F063D0.1442EBDD@dircon.co.uk>
Rendered body type: text/plain

Alexander Kurpiers wrote:

> >
> > - Charles
>
> I think you will have to do sequential decoding (i.e. Fano) for such a
> code. Using the Viterbi algorithm will need lots of memory and time.
>
> 73' Alexander
>
> --

I was going to use about a K=7 -> 10 code. Time is not very relavent. I am
looking at doing a broadcast system rather then ARQ and I am happy
to wait minutes for the message to be decoded. The ability to get the
message through is more important than decoding delay. I am also
considering concatenating the Convolutional code with a RS code
or even Golay encoding the data, interleaving it, convolutional coding
it then a final interleave. No CRC, any words that fail the golay
parity check code would be marked as bad (possibly using either a
block character or displaying them in another colour -> color).
As far as memory is concerned the DJGPP stuff allows virtual memory,
with a 4 GB limit I think. The real problem is making sure my interrupt
drivers are locked in memory and don't get swapped out! The
current software disables swapping and has the drivers written in C.
This limits me to the 64MB I have in my P.C.

All this is useless unless the preamble detect works well, (thats
another story, I will look back through the archives on that one, although
I am looking at a 13 bit Barker sequence with each data bit covering 4
transmitted bit periods. I could use a PN sequence I suppose.

Still don't hold your breath! I will concentrate on the analogue part
first. I have been looking at some of the work the EME people have
been doing on a DSP based receiver using the ADSP2181.
I see Johan KC7WW gets a mention!

- Charles G4GUO

21. (no subject)

From: Thomas Troike <ttroike@postoffice.worldnet.att.net>
Date: Tue, 12 Aug 1997 18:31:07 +0000
To: hfsig@tapr.org
Message-ID: <19970812183101.AAA17834@LOCALNAME>
Rendered body type: text/plain

Please, I wish to unsubscribe

>
>

22. Re: [HFSIG:2399] Re: Static Crashes and FEC

From: Antony Martin <anthony.martin@gecm.com>
Date: Wed, 13 Aug 1997 13:58:10 +0100
To: hfsig@tapr.org
Message-ID: <970813135810.2325@oas01.gmcl.co.uk>
Rendered body type: text/plain

Hi charles & all,

> As far as DSP noise blankers are concerned, Frerking has nothing
> on them. I seem to remember Antony that the Marconi DSP HF
> receiver doesn't have a noise blanker either?

No, I don't think it does. It has 4-channel ISB now though. There's
also an EER version of the drive for those 500kW valve transmitter
jobs. (That's Envelope Elimination & Restoration.)

> My original idea was to use a Xtal Oscillator to down convert to
> a '10 Khz IF', filter it then use a complex mixer within the DSP code to
> move the signal to the wanted frequency 'a sort of DDS in S/W'. That
> however makes the DSP code a lot more complex as I have to pass

> information from the host PC as to the receive frequency etc. It might
> be a better way of doing it though, even though I would have to sample
> at a much higher rate than at present, but using decimating filters after the
> frequency conversion I can get back to something slower.

Sounds OK. I do it all the time (I mean I don't, the DSPs do).
Sounds like you'd be better off with a dual-port RAM PC
interface.

'DDS in SW' is little more than a rotating phasor created by repeatedly
multiplying your rotating phasor by a fixed one. If you remember your
complex number theory, multiplying phasors adds the angle. Repeated
adding to the angle causes rotation. There are other methods too. You
have to do it anyway to frequency-shift incoming samples to correct for
the frequency error.

> If I sample fast enough then the time offset of the A/D should be
> insignificant as far as phase is concerned? I will check the Crystal
> Semiconductors CS4215 data sheet. I have a PDF copy from their WWW
> site. It is a 16 bit device and has a programmable gain input stage before
> the A/D which has a 20 db range in 1.5 db steps.

I'll try and get the DS too - it might be a useful device. I'm using
a pair of texas TLC320AC02, 14 bit codecs. They don't sample
simultaneously either. But they're only 12us apart but it's enough
to degrade the opposite sideband rejection by a number of dB at the
higher frequencies (ie. furthest from carrier). In theory I should
put some all-pass FIR filters in it to tweak the delay, but it's not
enough to cause a problem.

> Finally Donald, yes we are allowed to run 1W EIRP on the band, a
> friend of mine is working on a 1KW Class D amplifier for the band!
> Using power mosfets and switched mode PSU components.
> His aerial has about -48db gain! I will be using something a bit more
> modest (about 100W) into a Marconi T or possibly a loop ant.

> There is an annual prize for the most innovative piece of equipment
> on the band Sponsered by the RSGB and Datong Electronics, but I
> wouldn't be after that would I ....? After all I already qualify
> for a Listeners award having heard someone more than 8 km away.
> In fact I can claim a 32 Km endorsement to the award as well!

> - Charles G4GUO

That sounds interesting... There's a couple of offline switching PSUs
in the junk room, in varying states of dismantle, must be around 1500W.
Probably make an ideal 73kHz transmitter. (SMPS are good transmitters
anyway without modification.) It's one of the PSUs from the 20kW solid-
state transmitter (JORN), but appears to be configured for single-phase
mains. Would we win a prize if we stuck a DSP in front of the FET drivers
to make a 1500W PSK transmitter?

Your comment on K=32 is noted.
I worked out it would take 743,000 years to test every symbol on my
parallel modem!!! How on earth am I going to test it???

How do you measure EIRP at 73kHz? You'd have to be at least 8000km
away to be in the antenna far-field, is specifying EIRP reasonable?
What does a calibrated antenna look like at 73kHz?

Anthony

- --
A.N. Martin   anthony.martin@gecm.com
Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

23. Aw: [HFSIG:2399] HFSIG digest 385

From: "Paulus" <paulstra@xs4all.nl>
Date: Wed, 13 Aug 1997 17:34:53 +0200
To: <hfsig@tapr.org>
Message-ID: <199708131535.RAA18631@smtp1.xs4all.nl>
Rendered body type: text/plain

Hello all,

I noticed that the coherent-evm56 program form Johan KC7WW workes fine with

the ve2iq slowbpsk software. (have had only an qso with my self using the audio loopback hi)
The difference in frequency on the 800 hz, using the evm56 or an very stable 800 hz tone with slowbpsk modulation, was within 0,1-0,2 hz, so looks like ok to me.
I have an evm56k with 4 mh xtal, no ceramic resonator. (why did Motorola change that ?).

Hope to make soon an qso with ve2iq slowbpsk: Is the ms40 the most used mode, with 8 bits ?

Cheerio for now, de Paul, pa0ocd

paulstra@xs4all.nl

24. Re: [HFSIG:2405] Aw: HFSIG digest 385

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 13 Aug 1997 09:50:58 -0800
To: hfsig@tapr.org
Message-ID: <199708131651.JAA28639@peak.org>
Rendered body type: text/plain

Hello Paul,

>Hope to make soon an qso with ve2iq slowbpsk: Is the ms40 the most used
>mode, with 8 bits ?
>

The "standard" HF mode is MS25, meaning 25mS baud-time (or if you convert that to baud rate = 40 bps), also ASCII is used for starting off a QSO. Set up a batch file with this as the default for COHERENT.

If things get tough, let the other station know that you are changing to the first level of error-correcting code, ET-1, as he needs to re-sync. If the error rate makes things tough, change to the next-higher level, ET-2. The throughput goes down when you do this, but you will see remarkable print even with no audible signals, assuming of course that both stations are within 5 Hz and they have successfully synchronized. It also require careful attention to details when maintaining your equipment for long-term stability and accuracy for this kind of work - definately not a hit-and-run kind of operation.

It does not take you long to recognize a signal off the air as to whether its ASCII, ET-1, or ET-2. The old human perception faculty can do wonderful things.

Some further transatlantic contacts now; VE2IQ worked G3IRM yesterday with good print both ways. Who knows, we may have QSO with you soon also?

73's

--Johan

25. Re: [HFSIG:2406] Re: Aw: HFSIG digest 385

From: ADIDAS753@aol.com
Date: Wed, 13 Aug 1997 13:54:42 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <970813135440_721475615@emout05.mail.aol.com>
Rendered body type: text/plain

how do you get off this list?

26. Re: [HFSIG:2407] Re: Aw: HFSIG digest 385

From: Charles Brain <chbrain@dircon.co.uk>
Date: Wed, 13 Aug 1997 20:12:30 +0100
To: hfsig@tapr.org
Message-ID: <33F2071E.4EA1FF91@dircon.co.uk>
Rendered body type: text/plain

ADIDAS753@aol.com wrote:

> how do you get off this list?

You have to send money !

- Charles

27. Re: [HFSIG:2407] Re: Aw: HFSIG digest 385

From: "NIKOS J. FARSARIS" <njf@lissos.tsinet.gr>
Date: Wed, 13 Aug 1997 21:30:15 -0200
To: hfsig@tapr.org
Message-ID: <33F24387.7E52@lissos.tsinet.gr>
Rendered body type: text/plain

ADIDAS753@aol.com wrote:
>
> how do you get off this list?

You do not! it is a cool list!

28. unscribe please

From: chris <chris@newcomb.com>
Date: Thu, 14 Aug 97 07:43:35 -0400
To: hfsig@tapr.org
Message-ID: 11433596006887@newcomb.com
Rendered body type: text/plain

[Empty message body]

29. Re: [HFSIG:2410] unscribe please

From: Monte.Tremont@nextel.com (Monte Tremont)
Date: Thu, 14 Aug 1997 09:05:40 -0400
To: hfsig@tapr.org, chris <chris@newcomb.com>
Message-ID: <00051BA9.4178@nextel.com>
Rendered body type: text/plain

Still getting a dupe message from everyone that tries to unscribe.
Started ever since I unscribed. I have sent all messages back to
tapr.org as requested yet I still get 20-30 pieces of mail in a 24
hour period, it is now totally unwanted mail and is becoming more than
an annoyance .. my employer thinks so TOO !!!

_____ Reply Separator _____
Subject: [HFSIG:2410] unscribe please
Author: chris <chris@newcomb.com> at INTERNET
Date: 8/14/97 6:51 AM

30. Re: [HFSIG:2402] Re: Static Crashes and FEC

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Thu, 14 Aug 1997 10:21:22 -0400
To: hfsig@tapr.org
Message-ID: <199708141421.KAA22814@aus-a.mp.campus.mci.net>
Rendered body type: text/plain

>first. I have been looking at some of the work the EME people have
>been doing on a DSP based receiver using the ADSP2181.
>I see Johan KC7WW gets a mention!
>
>- Charles G4GUO

Charles,

I'd be interested in whatever you find that you can pass along on the
EME/DSP issue.

Don, W4DH

31. Re: [HFSIG:2412] Re: Static Crashes and FEC

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 14 Aug 1997 08:55:40 -0800
To: hfsig@tapr.org
Message-ID: <199708141555.IAA00987@peak.org>
Rendered body type: text/plain

Don,

I passed some info on these some time ago .... just in case anyone missed it:

>>first. I have been looking at some of the work the EME people have
>>been doing on a DSP based receiver using the ADSP2181.
>>I see Johan KC7WW gets a mention!
>>
>>- Charles G4GUO
>

>Charles,
>
>I'd be interested in whatever you find that you can pass along on the
>EME/DSP issue.
>
>Don, W4DH
>

QST July 1997, page 88. "Softradio: The Future Is Now." for the
GNOME-group's efforts. This is a microwave receiver using DSP in the second
and third IF. All I helped with was to find some obscure bug.
http://www.ao.com/~~gnome where you can see some pictures of the front panel
with FFT waterfall as used for weak signal work. That particular project
uses the EZ-KIT 2181 DSP. Bob Larkin W7PUA, used it to actually receive and
record the Mars mission test signal some while ago (there was some 10 people
world wide that succeeded).

The other project that seemed to have found a niche with EME experimentors,
is one that I wrote up for QEX September 1996, page 8: "A DSP-Based Audio
Signal Processor." It features some rather nice brickwall bandpass filters,
digital passband frequency shifing (a digital BFO), and LMS denoiser/Wiener
autonotcher - also based on the EZ-KIT. There was the usual spate of
inquiries shortly after the article appeared, then after about a year there
was another big flood of inquiries - guess the project was put to good use
and someone published something about it. The best application that I heard
about was that it was being evaluated by the folks at the Stanford satellite
building project for inclusion in one of their missions. That was quite an
honor.

--Johan

P.S. I have three spare complimentary copies of the QEX article with the
sound processor article - if anyone is interested, you're welcome to have
one. Please e-mail me.

32. Re: [HFSIG:2412] Re: Static Crashes and FEC

From: Antony Martin <anthony.martin@gecm.com>
Date: Thu, 14 Aug 1997 17:22:37 +0100
To: hfsig@tapr.org
Message-ID: <970814172237.16613@oas01.gmcl.co.uk>
Rendered body type: text/plain

Am I correct that a licensee without morse is not allowed to use bands
'below 30MHz' because international rules require morse ability on 'MF
and HF bands' (See the RA web site).

Does this suggest that class B licencees would be allowed to use LF?

Anthony

33. Kalman filter & Bitsync

From: zs6awk@global.co.za (Danie Brynard)
Date: Fri, 15 Aug 1997 06:54:57 +0200
To: hfsig@tapr.org
Message-ID: <199708150454.GAA26697@mail.global.co.za>
Rendered body type: text/plain

While reading about GPS and Deep space probes I noted that Kalman filters
are often mentioned.

1. How does these things work ?
2. Are there any literature explaining the basics in an 'amateur' way ?
3. Can we use it for synchronization in amateur modems ?

Danie

++
Danie Brynard   Email: zs6awk@global.co.za
PO Box 15133,Lyttelton,0140,South Africa
GRID LOC: KG44CE,   Tel/Fax: (2712) 6640330
++

34. Re: [HFSIG:2415] Kalman filter & Bitsync

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 15 Aug 1997 09:29:31 -0800
To: hfsig@tapr.org
Message-ID: <199708151629.JAA08912@peak.org>
Rendered body type: text/plain

Hello Danie,

>While reading about GPS and Deep space probes I noted that Kalman filters
>are often mentioned.
>
>1. How does these things work ?
>2. Are there any literature explaining the basics in an 'amateur' way ?
>3. Can we use it for synchronization in amateur modems ?

Interesting question.

Perhaps you remember some time ago we had the discussion about the Harris
patent for using a transversal filter to do channel equalization on embedded
training bits. RACAL advertises that they use a Kalman-directed decision
feedback equalizer in their "HF SkyFax" modems.
Working around patent issues or is this technical merit - thats the question
that remain.

--Johan

35. New Uploads for PSK31, MT63ASC

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 20 Aug 1997 12:02:13 -0800
To: dsp@tapr.org, HFSIG@tapr.org
Message-ID: <199708201903.MAA08235@peak.org>
Rendered body type: text/plain

Hi all,

Seems like some postings were lost during the time the server was down - so
here goes again (third time is perhaps lucky).

I have uploaded recent code releases that may be of general interest to the
DSP code repository. They will be available as soon as system admin moves
them to:

ftp://ftp.tapr.org/tapr/SIG/dsp/recent_uploads/mt63asc1.* and psk31_1.*

MT63ASC --- by Pawel Jalocha, SP9VRC: 63 tone,
            1000 Hz bandwidth, 630 bps modem.
            Very impressive performance.

PSK31   --- by Peter Martinez, G3PLX. This is the
            European PSK system. Works in very poor/low
            band conditions. Outstanding performance
            only 62.5 Hz bandwidth! Comes with excellent
            user interface.

Thanks to Pawel and Peter for these contributions - we are indebted to them
for permission to distribute these code packages: source code, loadable
files and instructions for you to see and explore.

Enjoy!

--Johan

36. Re: [HFSIG:2414] Re: Static Crashes and FEC

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 20 Aug 1997 21:41:53 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199708210441.VAA04675@servo.qualcomm.com>
Rendered body type: text/plain

>Am I correct that a licensee without morse is not allowed to use bands
>'below 30MHz' because international rules require morse ability on 'MF
>and HF bands' (See the RA web site).

That rule applies only to *amateur* licensees and amateur bands. I
don't know the legal status of the UK 73 KHz band, but in the US the
"lowfer" band (160-190KHz) is *not* an amateur band. Operation here is
permitted without any kind of license, though power levels and antenna
sizes are severely limited with respect to normal amateur levels.

The US military also uses that frequency band for the Ground Wave
Emergency Network (GWEN), a store-and-forward packet radio network
supposedly established to provide robust communications after a
nuclear attack for ordering a retaliatory strike. So like any unlicensed

user, you must tolerate any interference from these and other systems,
while not causing any interference of your own.

Phil

37. Re: [HFSIG:2415] Kalman filter & Bitsync

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 20 Aug 1997 22:09:29 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199708210509.WAA04880@servo.qualcomm.com>
Rendered body type: text/plain

Danie asked about Kalman filters.

While I won't pretend to know a lot about them, I can say a few basic
things about them.

A Kalman filter is a filter that attempts to estimate a modeled set of
state variables from a series of noisy observations. A good example is
a navigation system (like a GPS receiver) that computes position,
velocity, acceleration, heading and other variables from a series of
raw position fixes. Another example is the determination of a
satellite's orbital elements from a series of range or range-rate
measurements from a tracking station.

Like all real-world measurements the fixes are noisy (GPS selective
availability, ionospheric dispersion, ordinary receiver noise, etc) so
the filter has to smooth them out. It does this with a recursive
filter (in DSP terminology we'd call it an IIR filter). The genius of
the Kalman filter lies in its ability to estimate the noise level in
the measurements and change the responsiveness (the gains) of these
filters appropriately.

For example, if your position measurements are very noisy you don't
want your filtered position to jump all over the place, so the Kalman
filter automatically lowers the filter gains. But if your measurements
are clean, the Kalman filter cranks up the gains so it can respond
rapidly to true movements.

Now if you are processing a whole bunch of measurements after the
fact, you don't really need a Kalman filter; you can use classic
least-squares-fitting techniques and crunch them in one big batch
until the results stop improving. But if you want intermediate results
as you go, the Kalman filter is the way to go.

A good example of a state estimation process in communications is a
phase locked loop. It takes a noisy carrier input and produces a
continual estimate of the input carrier's true phase as a function of
time. Depending on the loop bandwidth, the PLL may respond rapidly to
actual changes in carrier phase (e.g., due to modulation or source
motion), or it may smooth the phase estimate over a relatively long
period, suppressing as much noise as possible.

Having said all that, I'm not sure how useful Kalman filtering would
be to us. One of the things computers make really easy is a posteriori
processing of packets, in which you buffer up the entire raw packet
before you actually begin to process it. This is a change from
traditional modem design where you extract carrier phase and
demodulate bits on the fly, with little or no memory in the
modem. E.g., if you're trying to estimate carrier phase at a certain
point in the packet you can do a better job by looking at "future"
samples, not just those in the past, and this is only possible by
buffering the whole packet before you start.

Phil

38. Using EVM interface to COHERENT BPSK

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 22 Aug 1997 08:31:48 -0800
To: DSP@tapr.org, HFSIG@tapr.org
Message-ID: <199708221532.IAA14998@peak.org>
Rendered body type: text/plain

Hi,

Those that are planning on using the PSK31 or Coherent DSP code, please take
note:

1) You will NOT be able to use these with the N1OWU FLASH-resident loader.

This is because these programs, or any other for that matter, that sets up their own reset/interrupt
vectors, will interfere with the LEONID/BIOS-style environment. Strange unexplained behavior
otherwise will result.

*****Use Doug's loader only with programs that conforms to Leonid/BIOS usage******

2) If you wish, you can place either of PSK31 or COHEVM in FLASH by following Doug's procedure
in FLSHBOOT. That really is very slick and you may use the flash programming program's default settings.
Using the OnCE debugger, load PSK31 (or COHEVM), then load FLSHBOOT, GO, wait till the debugger
regains control and you're done.

Good luck, 73

--Johan

39. Re: [HFSIG:2357] Re: SQPSK

From: David A Willmore <willmore@cig.mot.com>
Date: Fri, 22 Aug 1997 10:37:20 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <9708221037.ZM5502@whelk>
Rendered body type: text/plain

On Jul 25, 5:48pm, Phil Karn wrote:
> Subject: [HFSIG:2357] Re: SQPSK
> >Same thing happened to my reply to Phil, Dave. The message popped out here
> >unintentionally. I sure wish HFSIG didn't put itself in the reply address!
>
> Long time readers of this list will remember that I've complained
> repeatedly and insistently about the broken behavior of the list
> server, such as the bogus Reply-To: header, the deletion of other
> recipients from the To: and CC.lists, and the munging of the Subject
> line.
>
> But, sadly, the maintainer seems to consider these bugs to be features.
> And they've spread to other lists as well. Sigh.
>
> Phil
>-- End of excerpt from Phil Karn

(Sorry for the lag)

By far the worst is the munging of the subject line. I like to sort my mail by subject. With this subject line prefix, that's impossible. All I can sort them by is server order. Sure, I could work around this, but it's not worth the trouble. Sigh.

Are you going to the DCC this year?

Cheers,
David

40. Re: [HFSIG:2421] Re: SQPSK

From: David A Willmore <willmore@cig.mot.com>
Date: Fri, 22 Aug 1997 16:42:03 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <9708221642.ZM18949@whelk>
Rendered body type: text/plain

On Aug 22, 10:52am, David A Willmore wrote:
> Subject: [HFSIG:2421] Re: SQPSK
> On Jul 25, 5:48pm, Phil Karn wrote:
> > Subject: [HFSIG:2357] Re: SQPSK
> > >Same thing happened to my reply to Phil, Dave. The message popped out here
> > >unintentionally. I sure wish HFSIG didn't put itself in the reply address!
> >
> > Long time readers of this list will remember that I've complained
> > repeatedly and insistently about the broken behavior of the list
> > server, such as the bogus Reply-To: header, the deletion of other
> > recipients from the To: and CC.lists, and the munging of the Subject
> > line.

> >
> > But, sadly, the maintainer seems to consider these bugs to be features.
> > And they've spread to other lists as well. Sigh.
> >
> > Phil
> >-- End of excerpt from Phil Karn
>
>
> (Sorry for the lag)
>
> By far the worst is the munging of the subject line. I like to sort my mail
by
> subject. With this subject line prefix, that's impossible. All I can sort
> them by is server order. Sure, I could work around this, but it's not worth
> the trouble. Sigh.
>
> Are you going to the DCC this year?

First off, apologies to all for my inadvartant mail to the group. Secondly, I take back what I said about the mailing list. The most annoying feature isn't the subject line mangling, it's the improper use of reply-to as Phil stated. I guess I just had to see an example of it to believe. Sorry for doubting you, Phil.

OBHFSIG:

I don't post much to this group as I'm more interested in VHF+ signals work--Phil's ideas about new link layer protocols, etc., but I find the conversation on the list fascinating and most of it makes sense. I'm a computer E, hence I don't have the strongest analog background, but I'm at least not ignorant. :)

I've got some ideas that I'm going to try out, so you might see me back in a few months. Sorry for the interuption.

Cheers,
David N0YMV

41. EVM56K for sale

From: Kory Hamzeh <kory@avatar.com>
Date: Mon, 25 Aug 1997 10:10:42 -0700 (PDT)
To: dsp@tapr.org, hfsig@tapr.org
Message-ID: <Pine.BSF.3.96.970825101010.1633A-100000@ns1.avatar.com>
Rendered body type: text/plain

I have an EVM56K here which is gathering dust. It includes all of the docs and software. This was not through the TAPR purchase. I would like to get $100 + shipping for it. It has been socketed for the flash chip and I'll include the flash chip.

Thanks,
Kory
AC6RN

42. bpsk testing

From: Eduardo Jacob <edu@kender.es>
Date: Mon, 25 Aug 1997 22:45:15 -0100
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19970825224515.006c4ddc@kender.es>
Rendered body type: text/plain

I have (at least I think) succesfully setup the EVM for coherent and psk31. I am seeking schedules on Europe (I am located in Bilbao, in the North of Spain) for trying it, or at least for hearing.

Best regards.

Eduardo/EA2BAJ

43. Re: [HFSIG:2423] EVM56K for sale

From: Geoff Bagley <geoff@gcbagley.demon.co.uk>
Date: Mon, 25 Aug 1997 23:16:01 +0100
To: hfsig@tapr.org
Message-ID: <D2GgJBAhQgA0Ewru@gcbagley.demon.co.uk>
Rendered body type: text/plain

In message <Pine.BSF.3.96.970825101010.1633A-100000@ns1.avatar.com>,
Kory Hamzeh <kory@avatar.com> writes

>
>I have an EVM56K here which is gathering dust. It includes all of the docs
>and software. This was not through the TAPR purchase. I would like to get
>$100 + shipping for it. It has been socketed for the flash chip and I'll
>include the flash chip.
>
>Thanks,
>Kory
>AC6RN
>
>
>
>

Would you ship it to the UK ?

I have a brother in Florida who might help .

73
--
Geoff Bagley
G3FHL.

44. RE: [HFSIG:2417] New Uploads for PSK31, MT63ASC

From: jalocha@axdele.cern.ch
Date: Tue, 26 Aug 1997 17:19:10 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199708261519.RAA03094@dxmint.cern.ch>
Rendered body type: text/plain

>Thanks to Pawel and Peter for these contributions - we are indebted to them
>for permission to distribute these code packages: source code, loadable
>files and instructions for you to see and explore.

And I want to thank to all those hams who test my crapy assembler source
often without any instruction how to use it :-)
I wouldn't write many of the modes if there wasn't someone out there
who tries them out and finds out that they actually work.

Pawel

45. bpsk testing

From: Eduardo Jacob <edu@kender.es>
Date: Tue, 26 Aug 1997 21:29:37 -0100
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19970826212937.006c4130@kender.es>
Rendered body type: text/plain

I have been playing with PSK31 and I have make a (one, uno) QSO with Jabi
EA2ARU we used 28.800 USB and we are about 40 Km away one from the other. I
was very interesting to have a Voice VHF link as well. Even with that, the
success has been only partial, I could copy him but he couldn't copy me.

Several thougths you can comment about the experiment.

0th) It has been my first real experience of testing something like this.
Very cool getting Jabi's printing!!!!
1st) It's difficult to link two points at 40 km in mountainous areas with HF
2nd) The main (only?) problem is the tunning. It seems you have to been
very close in freq to be able to tune with the aids. The method we used (I
still don't know if it very repeatable is to tune by ear: I ear at Jabi's
monitor signal (by VHF) and tune to get the same tone) and after that, with
the preamble (TX-ON) and the RIT. Any advices to get the tunning?
3rd) In order to see how it works you can use the monitor signal of the
radio to inject your own signal in the modem and it starts printing.
4th) Sometimes the EVM program unexpectedly stops working I haven't find
which is the causing point.
5th) I don't know how much important is the input level. I can see that the
indicator varies when I move it, but I can see which is the rigth value.
6th) If some one make scheduled contacts I would be interested to try to
copy them.

Eduardo/EA2BAJ

46. Re: [HFSIG:2427] bpsk testing

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 26 Aug 1997 17:17:58 -0800
To: hfsig@tapr.org

Message-ID: <199708270018.RAA05975@peak.org>
Rendered body type: text/plain

Eduardo,

How exciting! A few comments and suggestions that may help.

1) While you are still finding your feet with these very narrow-band modes, it is a good idea to calibrate your transceiver against WWV or whatever standard you have. Get it down to within 5Hz or better - this is important. If both parties have done that, you can decide on a frequency to test on. Make sure both ends are using USB, or LSB as you wish. The dial the same frequency both ends. The PSK31 will start at 1000 Hz carrier frequencies for both RX and TX, so if you are on USB, your emitted carrier will be 1000 Hz above the displayed frequency, for LSB it will be 1000 Hz below. Makes sense?

2) Once you start transmitting, do not touch the transceiver as you will introduce so many variables that it will only add to the frustration . The program on the PC will allow you to work within a relative large passband and steer the DSP demodulator to resolve almost any frequency differences.

Press the net function (F3) key now - that will lock/unlock the RX and TX carrier frequencies so you can either move them both up or down in frequency, or alternatively to move only the receive and lock the transmitter in place. Let the your friend start transmitting, perhaps initially only the carrier (F5) - no typing in the beginning. If you look at the tuning indicator you should be able to zero the yellow dots to the bottom of the yellow bar using the Left and Right keyboard keys. You can tune to within 0.1 Hz - notice how much 0.1 Hz can move the phase dots! When you got that, let your friend start typing some characters on the keyboard and you should see the yello dot break apart in two patterns, one located on the top of the vertical bar and the other on the bottom. You probably will see perfect print now. Then reverse the procedue, this time let your friend tune up the same way.
You might also notice that on every time you reverse the transmission, you will note that there is a little bit of drift between the pair of you. Correct that with the Left/Right arrow keys, but do not touch your tranceiver - do it all from the keyboard.

As you gain experience, you will quickly be able to tell whether the station you are tuning to is higher or lower in frequency - at this time you probably want lock the RX/TX carriers (F3) and tune as close as you can. Then F3 to start "netting" - and you will be having a QSO in a few minutes.

PSK31's interface by G3PLX is very well done and layed out convenietly. The Coherent interface, I am afraid will take more than a couple of pages to explain. There are a lot of details there, but the key again is to start inititially within 5 Hz - otherwise it just won't print anything. Remember the "END" key to start synchronizing, and once you start seeing print, "ESC" to lock the parameters. Hit the 'TAB" key to send a good sinchronizing sequence - that allows others in the "round table" to synchronize. In PSK31, this is all automatic.

Regarding your EVM program locking up - we don't see that happen here, but if it happens during transmission, it may be a good idea to add some RFI measures; on the DC power, all the RS-232 lines and other interfacing wires like for PTT etc. I have bypass capacitors on everyting entering or leaving the metal enclosure, just for good measure.

Hope this helps - good luck with your experiments on the air, 73

--Johan.

>0th) It has been my first real experience of testing something like this.
>Very cool getting Jabi's printing!!!!
>1st) It's difficult to link two points at 40 km in mountainous areas with HF
>2nd) The main (only?) problem is the tunning. It seems you have to been
>very close in freq to be able to tune with the aids. The method we used (I
>still don't know if it very repeatable is to tune by ear: I ear at Jabi's
>monitor signal (by VHF) and tune to get the same tone) and after that, with
>the preamble (TX-ON) and the RIT. Any advices to get the tunning?
>3rd) In order to see how it works you can use the monitor signal of the

>radio to inject your own signal in the modem and it starts printing.
>4th) Sometimes the EVM program unexpectedly stops working I haven't find
>which is the causing point.
>5th) I don't know how much important is the input level. I can see that the
>indicator varies when I move it, but I can see which is the rigth value.
>6th) If some one make scheduled contacts I would be interested to try to
>copy them.
>
>   Eduardo/EA2BAJ
>
>

47. RE: [HFSIG:2427] bpsk testing

From: jalocha@axdele.cern.ch
Date: Wed, 27 Aug 1997 09:36:52 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199708270736.JAA12130@dxmint.cern.ch>
Rendered body type: text/plain

>4th) Sometimes the EVM program unexpectedly stops working I haven't find
>which is the causing point.

I see this with my setup too. The EVM can start giving out cracky noise
when I connect/disconnect the input cable or the radio antenna.
Switching audio cables from one radio to the other or the serial cable
between the two EVM connectors would crash the programs too.
I noticed same in the shack of our radio club, there moving the beam rotors
would as well cause program crashes. I remember someone's message about
the Crystal CODEC which can generate spurious interrupts when it gets
overvoltaged input. And one interrupt too much makes the data stream
from the CODEC desynchronized and the DSP gets the control words
as the audio data, etc. Possibly one could make the software to recover
but then the CODEC serving interrupts would take more time.

Anyway the cure for me was to connect together the ground of all my radios,
of the PC and of the EVM. The EVM ground connection is simple a crocodile
holding one of the screws of the serial connector.
This arrangement may cause ground loops in principle but it works for me.

Another possibly solution would be to put diode protection on the CODEC
input/output and make good shielding, etc.

Pawel

September 1997

Original source file: hfsig.9709
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9709

1. HF BPSK Progress

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 01 Sep 1997 16:29:16 -0800
To: DSP@tapr.org, HFSIG@tapr.org
Message-ID: <199709012330.QAA28935@peak.org>
Rendered body type: text/plain

Hi all,

A few tidbits of news on the status of HF BPSK:

1) I have received a note from Peter, G3PLX, that EA2BAJ and EA2ARU (Jabi and Eduardo) joined the 7MHz PSK31. Congratulations - I know Eduardo and Jabi have been looking for a QSO for some time. Seems like the PSK31 activity keeps on growing in Europe.

2) Paul, PA0OCD, and Bill, VE2IQ had a transatlantic contact on 20 meters using Coherent (ET-1). The remarkable thing is that Paul managed this with an indoor loop antenna!

3) Bill, WB6RIJ and myself (KC7WW), had a very nice QSO on 30 meters using MT63 over a distance of about 1500 miles. We were really impressed that the copy was 100% even after I turned my power all the down as far as I could - probably a few milliwats. Bill was running only 2Watts. MT63 appears to have outstanding capabilities. Thanks to Pawel, SP9VRC, for this wonderful contribution.

If anyone is interested in joining in, 7081 (carrier) in the evenings is a good place to listen.

73's

--Johan

2. Aw: [HFSIG:2430] HFSIG digest 397

From: "Paulus" <paulstra@xs4all.nl>
Date: Tue, 2 Sep 1997 07:28:10 +0200
To: <hfsig@tapr.org>
Message-ID: <199709020529.HAA07156@smtp2.xs4all.nl>
Rendered body type: text/plain

Hello All,

Last night I had an qso with AA1A, Dave and VE2IQ, Bill running Coherent slowbpsk ms25 mode both on ET1 And ET2 mode.
Both the evm56 program Cohevm11 as well the orignal sigma/deltabord with an home brew bpsk modulator, worked well.
The signals could sometimes be heard, most of the times barely not (in 500 and 100 hz BW).
It is of absolute importacne to tune in very accurate, which can be done running FFTZZ and/or Tune.com from Bill, ve2iq.

I have an old FT-107 with digital memory and changed only 1 resistor in the Clarifier circuit, so can tune on 1 hz or better, very easy.
The stability is good enough , wenn you keep watch of the indication in the ve2iq program. I ran version 6.7.

Yes hope soon to make an transatlantic qso on the other modes: Psk31 and MT63asc.
Perhaps it is better to be 1 khz down from the 14.081 ? because then we do not interfere then with Coherent signals , which often are there, but could not be heard hi.

Cheerio for now, and I try to be very often round the 14.081 at 20.00 UTC.

73 de Paul, pa0ocd

paulstra@xs4all.nl

3. Re: [HFSIG:2430] HF BPSK Progress

From: Eduardo Jacob <edu@kender.es>
Date: Tue, 02 Sep 1997 12:49:36 -0100
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19970902124936.006c7058@kender.es>
Rendered body type: text/plain

>1) I have received a note from Peter, G3PLX, that EA2BAJ and EA2ARU (Jabi
and Eduardo)

Well, Eduardo and Jabi ;-)

Eduardo/EA2BAJ

4. TAPR EVM interface boards

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 02 Sep 1997 12:46:58 -0800
To: DSP@tapr.org, HFSIG@tapr.org
Message-ID: <199709021948.MAA02973@peak.org>
Rendered body type: text/plain

Hi All,

I have recently been getting a number of e-mails from enthuisiasts asking
about the proposed TAPR EVM interface kit - this is what I can tell you
about it:

I am helping out with testing a prototype, which looks quite good. The kit
consists of two small PCBs - one goes on the front panel and the other fits
on the rear panel. Connections are made to the EVM using ribbon cable and a
couple of soldered fly leads. You do not need to solder anything onto the
EVM, there is an 16-pin IDC header that plugs into J7. The intention is to
use the DSP-93 enclosure for which new silk-screened front/rear panels are
provided.

This is basically all that I know at the moment - please direct further
questions about availability and pricing to TAPR.

Thanks and 73's,

--Johan, KC7WW

5. EVM boards

From: "Edson W.R. Pereira" <ewp@usa.healthnet.org>
Date: Wed, 3 Sep 1997 10:15:22 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <Pine.SUN.3.95.970903101227.10619N-100000@satellife.healthnet.org>
Rendered body type: text/plain

I'm having problems locating a supplier for the DSP56002EVM. I've got 3
representatives from motorola and all can only supply the board in 5 to 6
weeks. Does any one know a place where I could purchase the board?

Thanks in advance for any info.

73,

Edson, N1VTN

6. Unsubscribe

From: Ed Kaprocki <kaprocki@iag.net>
Date: Wed, 3 Sep 1997 21:13:58 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BCB8AE.721DB0E0@d069.aspg.fl.iag.net>
Rendered body type: text/plain

[Empty message body]

7. new url

From: "Robson" <robson@aresta.org>
Date: Thu, 4 Sep 1997 16:44:12 -0300
To: "Wolf" <wks@netalpha.com.br>
Cc: "aresta na usa.net" <aresta@usa.net>,
"CD-ROM Revista Editora Europa Aydano Roriz" <suporte.revista.cd-rom@sti.com.br>,
"FATORGIS revista" <fator.gis@sul.com.br>, <hfsig@tapr.org>,
"Ivan Pereira de Mello" <stmello@br.homeshopping.com.br>,
"Juan Luis Mendez" <juancho@alphanet.com.br>,
"Ruddi" <ruddi@alphanet.com.br>
Message-ID: <199709041940.QAA00102@netra01.origin.com.br>

Rendered body type: text/plain

meu novo e-mail:

robson@aresta.org

web:

http://www.aresta.org

abraços

robson

8. BPSK and MT63 Testing

From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
Date: Fri, 05 Sep 1997 17:03:56 +0200
To: hfsig@tapr.org
Message-ID: <3.0.3.32.19970905170356.006b283c@bicc00.bi.ehu.es>
Rendered body type: text/plain

We have been testing MT63ASC and it seems to work well. We have tested on
10 meters. The signal is acquired very fast. We still have to take
meassures in order to send them to pawel. Perhaps I am missing something,
but does the modem in it's present state control any (software) handshake
protocol?

The next thing to try is coherent, which we expect to do this weekend.

Eduardo

Eduardo Jacob - Area de Ingenier'ia Telem'atica
Departamento de Electr'onica y Telecomunicaciones
ETSII y de IT Tel: +34-(9)4-427 8055
UPV / EHU Fax: +34-(9)4-441 4041
Alda Urquijo s/n E-mail: jtpjatae@bi.ehu.es
E-48013 - Bilbao (Spain) : edu@kender.es
Ham: EA2BAJ VHF PACKET: EA2BAJ @ EA2URV
PGP Key available: Send email w/subject: PGPKEY
fingerprint : D8 FA 54 49 F7 40 BA AA 09 1A 73 40 A4 26 ED BD

9. RE: [HFSIG:2437] BPSK and MT63 Testing

From: jalocha@axdele.cern.ch
Date: Sat, 6 Sep 1997 08:25:23 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199709060625.IAA17082@dxmint.cern.ch>
Rendered body type: text/plain

Hello Eduardo,

> We have been testing MT63ASC and it seems to work well. We have tested on
>10 meters. The signal is acquired very fast. We still have to take
>meassures in order to send them to pawel. Perhaps I am missing something,
>but does the modem in it's present state control any (software) handshake
>protocol?

There is NO handshake. Whatever you send in (by typing on the keyboard)
is being transmitted or stored in a buffer when the transmitter is off
or busy sending previous characters. When the buffer overflows the modem
sends back BELL characters in response to every user character which does
not fit. The buffer size is 256 characters and can be easily changed
(look into the parameters at the top of the source file).

As well, there is no handshake between the communicating stations,
no ARQ or anything.
The modem works like an RTTY terminal, it just sends the data out
and doesn't care if the listening station(s) is there at all :-)

The best modem's measure/comparison is to start up another digital mode
and see how much power you need to achieve similar bit/character error rate
in same HF conditions. Then compute the ratio:

RF power
---
effective user bit rate

for each mode.
- this will be directly the energy per bit you need to achieve given

error rate. MT63ASC is 70 bits/sec thus if you use say 35 W than you need
35/70 = 0.5 Joules/bit.

When you measure this for several modes like RTTY, AMTOR, PACTOR, PSK31,
MT63ASC, COHERENT, etc. and you send the measurements to the list you
may trigger a long-lasting discussion :-)

Pawel

ps. I will be out of e-mail for next 10 days so if you have more questions
    they will have to wait.

10. Re: [DSP:645] TAPR EVM interface boards

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Sat, 6 Sep 1997 12:35:47 -0500
To: dsp@tapr.org, "HF SIG list   mailing"<hfsig@tapr.org>
Message-ID: <v03110703b03744c35b1d@[208.134.134.40]>
Rendered body type: text/plain

>Hi All,
>
>I have recently been getting a number of e-mails from enthuisiasts asking
>about the proposed TAPR EVM interface kit - this is what I can tell you
>about it:
>
>I am helping out with testing a prototype, which looks quite good. The kit
>consists of two small PCBs - one goes on the front panel and the other fits
>on the rear panel. Connections are made to the EVM using ribbon cable and a
>couple of soldered fly leads. You do not need to solder anything onto the
>EVM, there is an 16-pin IDC header that plugs into J7. The intention is to
>use the DSP-93 enclosure for which new silk-screened front/rear panels are
>provided.
>
>
>This is basically all that I know at the moment - please direct further
>questions about availability and pricing to TAPR.

I think that is about all we have at this point. We wilkl not have a date
or price until we get a little further along.

Cheers - Greg

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd

    "Before you become too entranced with gorgeous gadgets
     and mesmerizing video displays, let me remind you that
     information is not knowledge, knowledge is not wisdom,
     and wisdom is not foresight. Each grows out of the other
     and we need them all."   -Arthur C. Clarke

11. Re: [HFSIG:2428] Re: bpsk testing

From: zs6awk@global.co.za (Danie Brynard)
Date: Fri, 05 Sep 1997 19:38:23 +0200
To: hfsig@tapr.org
Message-ID: <199709071035.MAA01200@mail.global.co.za>
Rendered body type: text/plain

BTW the bpsk users, a comment:

My hf rig (TS690S) provides +-10ppm stability which means a freq error of
+-300Hz at 30MHz. Although the rig's resolution is 1Hz it is quite
'unstable' for psk work ?

Danie

++
Danie Brynard   Email: zs6awk@global.co.za
PO Box 15133,Lyttelton,0140,South Africa
GRID LOC: KG44CE,   Tel/Fax: (2712) 6640330
++

12. Re: [HFSIG:2440] Re: bpsk testing

From: forrerj@peak.org (Johan Forrer)
Date: Sun, 07 Sep 1997 08:03:11 -0800
To: hfsig@tapr.org
Message-ID: <199709071504.IAA13767@peak.org>
Rendered body type: text/plain

Danie,

If you can calibrate your dial against WWV, and your tranceiver's frequency
generation is based on a controlled single reference, you are in good shape.
A number of people uses the ICOM 706 with excellent results when fitted with
a temperature-stabilized oscillator.

I have an older ICOM 751 - it has just too many individual crystals that
generate frequencies. Even with the high stab. oscillator, it walks around
all over the place. I can, however, set it to within a few Hz accuracy by
going calibrating it to WWV. Both PSK31 and Coherent gives you a frequency
calibrator that allows one to tune to within one Hz.

--Johan

>BTW the bpsk users, a comment:
>
>My hf rig (TS690S) provides +-10ppm stability which means a freq error of
>+-300Hz at 30MHz. Although the rig's resolution is 1Hz it is quite
>'unstable' for psk work ?
>
>Danie
>
>++
>Danie Brynard   Email: zs6awk@global.co.za
>PO Box 15133,Lyttelton,0140,South Africa
>GRID LOC: KG44CE,   Tel/Fax: (2712) 6640330
>++
>
>

13. dsp56001

From: Stephen PEARCE <spearce@ccu1.auckland.ac.nz>
Date: Mon, 8 Sep 1997 12:35:40 +1200 (NZT)
To: hfsig@tapr.org
Message-ID: <199709080035.MAA28967@ccu1.auckland.ac.nz>
Rendered body type: text/plain

>   I have a dsp56001 adb and sigma delta adc/dac board
> to go with it. I would like to become involved
> in the coherent bpsk experimenting currently underway and was told that
> there is a British hf sig contributor who has
> a similar set up. Unfortunately I have lost the software loader that goes with this
> board .. and was wondering if you could put me in touch with him or anyone else who
> might have the info needed for me to start programming this board using the
> adapter card that goes with it for the pc
>
>   Thanks
>   Stephen Pearce
>   zl1any
>

14. Possible QRM

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 08 Sep 1997 22:12:07 -0800
To: HFSIG@tapr.org
Message-ID: <199709090513.WAA20288@peak.org>
Rendered body type: text/plain

Hi all,

I saw this rather disturbing message a little earlier. Just in case someone
is inadvertendly causing interference to these folks.

I know for a fact that this is not caused by G3PLX as he is out of town at
the moment and second that his program does not include a beacon mode. In my
estimation, this signal appears to originate from somewhere in Europe by
someone using a version of the PSK31 code that has the beacon mode. Perhaps
forgot to change the setup string it seems.

What is puzzling is that it takes extreme efforts to get to within a Hz of

the operating frequency and also knowing that there is a sked going on at
that time sounds like perhaps more than a coincidence?

Thanks,

--Johan, KC7WW

X-Authentication-Warning: ns3.qth.net: majordom set sender to
owner-bpsk@qth.net using -f
X-Sender: bill@ietc.ca
Date: Mon, 08 Sep 1997 18:38:22 -0400
To: bpsk@qth.net
From: Bill de Carle <bill@ietc.ca>
Subject: [BPSK] frequency change
Cc: Peter.Martinez@btinternet.com
Sender: owner-bpsk@qth.net
Precedence: bulk
Reply-To: Bill de Carle <bill@ietc.ca>

Hi everyone:

As you know, we have been having excellent transatlantic skeds every day
at 2000 GMT on 14.081 for the last month or so.

Today G3IRM and I were in the middle of a QSO when a station identifying
itself as G3PLX (in CW) apparently decided to start up a PSK31 beacon on
that freq. The transmissions consisted of a period of PSK31 followed by a
CW ident, repeated endlessly.

Needless to say, our C-BPSK QSO was over.

I don't know if it was really Peter (G3PLX) or not, but since both G3IRM
and I had no difficulty copying his ident we are certain that is the
callsign that was being used. I tried calling him directly on CW but he
would not answer.

I discussed this with G3IRM by telephone and we feel it would be best to
change our sked and move to 14.082 henceforth.

We'll still meet on 14.081 for the next day or so to make sure everyone
knows of the sked change, then we will move formally to 14.082.

Thanks for your understanding,

73 de Bill VE2IQ

Submissions: bpsk@qth.net
Subscription/removal: bpsk-request@qth.net

15. Matlab usage

From: RLANIER@mailb.harris.com (RLANIER)
Date: Wed, 10 Sep 1997 14:15:22 -0400
To: hfsig@tapr.org
Message-ID: <416e3a70@mailb.harris.com>
Rendered body type: text/plain

HELP!!!

I'm trying to learn to use Matlab WITHOUT a user's guide or manual.
Does anyone know of a web site that lists some helpful hints on how to
use Matlab?

Any help would be appreciated.

Tony

16. Re: [HFSIG:2444] Matlab usage

From: Keith McElveen <jkm@mclean.sparta.com>
Date: Wed, 10 Sep 1997 15:44:21 -0400
To: hfsig@tapr.org
Message-ID: <3416F895.7054@mclean.sparta.com>
Rendered body type: text/plain

I would suggest

www.mathworks.com

which is the company's website. You might do a search on "matlab" with
your favorite search engine. I don't particularly have a favorite but
have found useful code by search for, for example, "matlab
communications" or some such. Good luck.

Keith McElveen
Sparta, Inc.
jkm@mclean.sparta.com

17. RE: [HFSIG:2444] Matlab usage

From: "Al Quiroga" <ea1bk@amsat.org>
Date: Thu, 11 Sep 1997 16:45:29 +0200
To: <hfsig@tapr.org>
Message-ID: <199709111649.RAA19976@scugat7.abaforum.es>
Rendered body type: text/plain

look for

http://www.powells.com/

73 de Al, EA1BK
ea1bk@amsat.org

> De: RLANIER <RLANIER@mailb.harris.com>
> A: hfsig@tapr.org
> Asunto: [HFSIG:2444] Matlab usage
> Fecha: miércoles 10 de septiembre de 1997 20:22
>
>   HELP!!!
>
>   I'm trying to learn to use Matlab WITHOUT a user's guide or manual.
>   Does anyone know of a web site that lists some helpful hints on how
to
>   use Matlab?
>
>   Any help would be appreciated.
>
>   Tony
>
>

18. digital test

From: Arvydas Vaisnys <vaisnys@zorba.jpl.nasa.gov>
Date: Thu, 11 Sep 1997 09:31:02 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19970911093102.006adf54@zorba.jpl.nasa.gov>
Rendered body type: text/plain

The VOA/JPL HF Digital Project is planning a test transmission on Friday
September 12, from Delano CA on a 75 degree Az toward Washington DC. A 45
minute test sequence will be repeated from 16:00 UT to 18:30 UT on 15.24
MHz, then from 18:30 UT till 21:00 UT on 11.965 MHz.

Each test sequence will consist of short segments of different modulation,
such as BPSK up through 32PSK, 16 and 64 QAM, at 8kbaud in a 10kHz channel
as well as unmodulated carrier and DSB AM. The purpose of this test is to
see how well these modulation schemes will work on a Class C transmitter.
It would be interesting to get some feedback from various places on factors
such as signal strength and adjacent channel interference. We would like to
hear from anyone who receives this signal by e-mail to
vaisnys@zorba.jpl.nasa.gov.

This test is contingent upon the successful integration of the modulation
equipment with the transmitter at Delano. If this cannot be done on time,
then this test will be postponed and unfortunately there may not be enough
time to send out a cancellation message.

Arv

19. Re: [HFSIG:2447] digital test

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 11 Sep 1997 12:46:50 -0800
To: hfsig@tapr.org
Message-ID: <199709111948.MAA24834@peak.org>
Rendered body type: text/plain

Arv,

Sounds interesting, just curious; what kind of receiver will handle a 10kHz bandwidth? Looks like something special, however, I'd be interested in participating - drop me an e-mail if you wish.

--Johan

>The VOA/JPL HF Digital Project is planning a test transmission on Friday
>September 12, from Delano CA on a 75 degree Az toward Washington DC. A 45
>minute test sequence will be repeated from 16:00 UT to 18:30 UT on 15.24
>MHz, then from 18:30 UT till 21:00 UT on 11.965 MHz.
>
>Each test sequence will consist of short segments of different modulation,
>such as BPSK up through 32PSK, 16 and 64 QAM, at 8kbaud in a 10kHz channel
>as well as unmodulated carrier and DSB AM. The purpose of this test is to
>see how well these modulation schemes will work on a Class C transmitter.
>It would be interesting to get some feedback from various places on factors
>such as signal strength and adjacent channel interference. We would like to
>hear from anyone who receives this signal by e-mail to
>vaisnys@zorba.jpl.nasa.gov.
>
>This test is contingent upon the successful integration of the modulation
>equipment with the transmitter at Delano. If this cannot be done on time,
>then this test will be postponed and unfortunately there may not be enough
>time to send out a cancellation message.
>
>Arv
>
>

20. New PSK31 from G3PLX uploaded

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 11 Sep 1997 12:56:17 -0800
To: HFSIG@tapr.org, DSP@tapr.org
Message-ID: <199709111957.MAA25789@peak.org>
Rendered body type: text/plain

Hi all,

I have uploaded the latest release of the G3PLX PSK31 code package to the DSP\recent_uploads directory. Look for PSK312.ZIP.

This package contains some very readable and informative technical text on HF PSK that I would recommend studying. Otherwise, note that you should not mix this code with previous releases of PSK31 - keep these seperate as these are not interchangeable!

Thanks to Peter for this contribution - have fun and let us have some reports on what you think.

73's

--Johan

PS. Tim (AA5DF), could you kindly move the code package from "upload" to "recent_uploads". Thanks much. J.

21. Re: digital test

From: Alan Bloom <abloom@manta.sr.hp.com>
Date: Thu, 11 Sep 1997 14:35:29 -0800 (PDT)
To: hfsig@tapr.org
Message-ID: <199709112135.AA064363729@manta.sr.hp.com>
Rendered body type: text/plain

Arvydas Vaisnys <vaisnys@zorba.jpl.nasa.gov> wrote:

>The VOA/JPL HF Digital Project is planning a test transmission
...
>Each test sequence will consist of short segments of different modulation,
>such as BPSK up through 32PSK, 16 and 64 QAM, at 8kbaud in a 10kHz channel
>as well as unmodulated carrier and DSB AM. The purpose of this test is to

>see how well these modulation schemes will work on a Class C transmitter.

I don't see how a class C transmitter will work with QAM (or DSB AM
for that matter). You will likely get huge symbol errors as well as
horrible adjacent-channel interference.

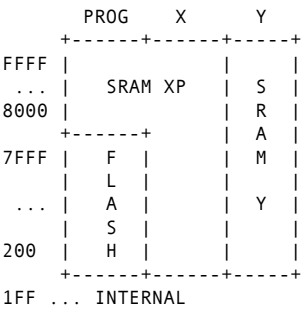
Unfortunately, I will be at work when the transmissions take place so
I can't listen in.

Alan Bloom N1AL

22. Extended memory for 56002EVM

From: David A Willmore <willmore@cig.mot.com>
Date: Wed, 17 Sep 1997 14:31:02 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <9709171431.ZM7200@whelk>
Rendered body type: text/plain

Is there any interest in a board that would give the 56002EVM 128K words of
memory? It would be three MCM6226-20's on a little daughter card that would
plug into the jacks over the 56002, the three SRAM and the FLASH chip. One
guess at a memory map would be:



The layout and schematic would be GPL'd, but I'm not sure if I could make
boards. If I did, they wouldn't be the prettiest things on the planet.

If anyone has any ideas, please let me know.

Cheers,
David

--

23. Re: [HFSIG:2451] Extended memory for 56002EVM

From: "Ulrich G. Schaarschmidt" <ulrich.schaarschmidt@UniBw-Hamburg.DE>
Date: Thu, 18 Sep 1997 09:46:51 +0000
To: hfsig@tapr.org
Message-ID: <199709180748.JAA08858@rzsparc-3.unibw-hamburg.de>
Rendered body type: text/plain

> The layout and schematic would be GPL'd, but I'm not sure if I could make
> boards. If I did, they wouldn't be the prettiest things on the planet.
>
> If anyone has any ideas, please let me know.
>
> Cheers,
> David
>
> --
Hello EVM-Fans,
if You like it, I can scan an article on this topic including
schematics (but article in german) an mail it to You. They are using
"old" / cheap cache-Rams from a PC (6 pieces of
MT5C2568/CY7C199/UM61256AK times 6) and a 74ACT139
for X/Y-adressing.

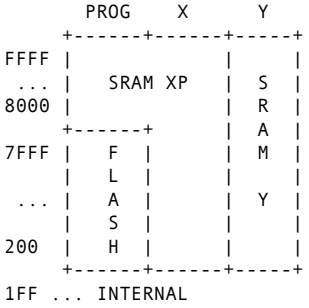
Best Greetings
de Uli, DJ1HM (@amsat.org)
Dr.-Ing. Ulrich G. Schaarschmidt
University of the Federal Armed Forces Germany
22039 Hamburg

24. RE: [HFSIG:2451] Extended memory for 56002EVM

From: jalocha@axdele.cern.ch
Date: Thu, 18 Sep 1997 10:46:25 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199709180846.KAA24968@dxmint.cern.ch>
Rendered body type: text/plain

>Is there any interest in a board that would give the 56002EVM 128K words of
>memory? It would be three MCM6226-20's on a little daughter card that would
>plug into the jacks over the 56002, the three SRAM and the FLASH chip. One
>guess at a memory map would be:

I could do long FFTs, deep interleaves and put more tones at lower baud rates
on the HF air :-)



Flash is better placed from 8000-FFFF, otherwise it is fine.
What about the on-board SRAM ?

Pawel

25. Re: [HFSIG:2451] Extended memory for 56002EVM

From: Charles Brain <chbrain@dircon.co.uk>
Date: Thu, 18 Sep 1997 12:42:58 +0100
To: hfsig@tapr.org
Message-ID: <342113C2.DF595BF2@dircon.co.uk>
Rendered body type: text/plain

David A Willmore wrote:

> Is there any interest in a board that would give the 56002EVM 128K words of
> memory? It would be three MCM6226-20's on a little daughter card that would
>
> If anyone has any ideas, please let me know.
>
> Cheers,
> David
>
> --

Yes that sounds interesting,

I hope to produce a DDS piggyback board for the EVM for my
digital radio experiments. I havn't decided how it will interface
yet but please bear in mind other expansion possibilities.
I.E don't preclude the addition of other boards as well.

- Charles

26. Re: [HFSIG:2454] Re: Extended memory for 56002EVM

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 18 Sep 1997 08:33:39 -0800
To: hfsig@tapr.org
Message-ID: <199709181534.IAA28587@peak.org>
Rendered body type: text/plain

David,

>David A Willmore wrote:
>
>> Is there any interest in a board that would give the 56002EVM 128K words of
>> memory? It would be three MCM6226-20's on a little daughter card that would
>>
>> If anyone has any ideas, please let me know.
>>
>> Cheers,
>> David

>>
>> --

I like the idea and would encourage you to go ahead.

You may just want to check out the design that I have posted on the TAPR web page that gives you 32KW of P and 16KW of each X/Y. I have been using it for about a year now and quite happy with it, however, there is a snag - with timing as tight as it is, you probably need to look very closely to any form of address decoding that you use - I use a 7ns GAL and still have to run the external memory with one wait state - which you want to avoid. Also things gets even worse as the clock speed goes to 80MHz.

I have seen a German design where he used 128K SRAM chips (three of them) with no address decoding. This runs with zero wait states. If you are interested in looking at this one, e-mail me for further details.

There are wonderful things one can do with a bit of memory.

73

--Johan

27. Re: [HFSIG:2452] Re: Extended memory for 56002EVM

From: David A Willmore <willmore@cig.mot.com>
Date: Thu, 18 Sep 1997 10:41:44 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <9709181041.ZM2464@whelk>
Rendered body type: text/plain

On Sep 18, 2:52am, Ulrich G. Schaarschmidt wrote:
> Hello EVM-Fans,
> if You like it, I can scan an article on this topic including
> schematics (but article in german) an mail it to You. They are using
> "old" / cheap cache-Rams from a PC (6 pieces of
> MT5C2568/CY7C199/UM61256AK times 6) and a 74ACT139
> for X/Y-adressing.

I think having that available would be great. Those should be very easy for people to find all around the world. Who can't find an old 486 MB to gut?

Cheers,
David

28. Re: [HFSIG:2453] RE: Extended memory for 56002EVM

From: David A Willmore <willmore@cig.mot.com>
Date: Thu, 18 Sep 1997 10:46:09 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <9709181046.ZM2590@whelk>
Rendered body type: text/plain

On Sep 18, 3:53am, jalocha@axdele.cern.ch wrote:
> I could do long FFTs, deep interleaves and put more tones at lower baud rates
> on the HF air :-)
>
> PROG X Y
> +------+------+-----+
> FFFF | | |
> ... | SRAM XP | S |
> 8000 | | R |
> +------+ | A |
> 7FFF | F | | M |
> | L | | |
> ... | A | | Y |
> | S | | |
> 200 | H | | |
> +------+------+-----+
> 1FF ... INTERNAL
>
> Flash is better placed from 8000-FFFF, otherwise it is fine.
> What about the on-board SRAM ?

That's just the reason that I posted the memory map.

My intention had been to replace the onboard SRAM as the way it's wired, uhhh, conflicts with my view of asthetics. Not to mention that there isn't a point in doing so as the daughter card would fill the address space--there isn't any other place to put that memory.

To move flash, I'll have to have people do a bit more soldering on the EVM itself to allow me to remap that memory. Is there any value in having Prog and X share that 32K (+/-) chunk? I haven't looked at any of the code that exists, so I don't know what coders prefer.

Cheers,
David

29. Re: [HFSIG:2455] Re: Extended memory for 56002EVM

From: David A Willmore <willmore@cig.mot.com>
Date: Thu, 18 Sep 1997 13:16:10 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <9709181316.ZM5177@whelk>
Rendered body type: text/plain

On Sep 18, 10:40am, Johan Forrer wrote:
> I like the idea and would encourage you to go ahead.
>
> You may just want to check out the design that I have posted on the TAPR web
> page that gives you 32KW of P and 16KW of each X/Y. I have been using it for
> about a year now and quite happy with it, however, there is a snag - with
> timing as tight as it is, you probably need to look very closely to any form
> of address decoding that you use - I use a 7ns GAL and still have to run the
> external memory with one wait state - which you want to avoid. Also things
> gets even worse as the clock speed goes to 80MHz.
>
> I have seen a German design where he used 128K SRAM chips (three of them)
> with no address decoding. This runs with zero wait states. If you are
> interested in looking at this one, e-mail me for further details.
>
> There are wonderful things one can do with a bit of memory.

6226s are 128Kx8, so three of them would provide a full 128K word of address space. I plan on using the top address bit to be the X/~Y select. I'll have to do some very simple decoding to allow the P bank to overlap the X bank, but nothing too complex--the eval board does the same trick, somewhat. No PAL, just, maybe an AC00 and an AC04. Should be no biggie--certainly no more than 3ns or so. I do intend to require that the onboard SRAM be disabled--some very minor surgery. You can cut a trace, lift a pin, or just suck the things right off the board, we don't need 'em. :)

I like this design because it's *so* simple. It would be even easier to no share SRAM between X and P. Would that be horribly bad? I guess that would leave the chip with only FLASH and the on-chip SRAM.

Cheers,
David

30. Re: [HFSIG:2453] RE: Extended memory for 56002EVM

From: David A Willmore <willmore@cig.mot.com>
Date: Fri, 19 Sep 1997 11:51:10 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <9709191151.ZM28948@whelk>
Rendered body type: text/plain

> I could do long FFTs, deep interleaves and put more tones at lower baud rates
> on the HF air :-)
>
>   PROG   X   Y
>   +------+------+-----+
> FFFF | | |
> ... | SRAM XP | S |
> 8000 | | R |
>   +------+ | A |
> 7FFF | F | | M |
> | L | | |
> ... | A | | Y |
> | S | | |
> 200 | H | | |
>   +------+------+-----+
> 1FF ... INTERNAL
>
> Flash is better placed from 8000-FFFF, otherwise it is fine.
> What about the on-board SRAM ?

Ok, this is my second take at your message. I spent some of last night staring at this map and you're right, I was confised when I wrote it. Here's the corrected version:

```
       PROG    X      Y
     +------+------+-----+
7FFF |   F  | I/O  |     |
     |   L  +------+  S  |
 ... |   A  |      |  R  |
     |   S  |      |  A  |
8000 |   H  |      |  M  |
     +------+      |     |
7FFF |             |  Y  |
 ... |  SRAM X/P   |     |
 200 |             |     |
     +------+------+-----+
1FF ... INTERNAL
000
```

This looks like it's going to be easy to decode--I'm crunching the K-maps in my spare time.

More when it's available....

Cheers,
David

31. Re: [HFSIG:2459] RE: Extended memory for 56002EVM

From: ADIDAS753@aol.com
Date: Sat, 20 Sep 1997 22:01:29 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <970920220128_1896458237@emout10.mail.aol.com>
Rendered body type: text/plain

how do i sign off this

32. Re: [HFSIG:2450] Re: digital test

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Sat, 20 Sep 1997 22:09:34 -0600 (MDT)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.3.96.970920220610.908C-100000@murdrum>
Rendered body type: text/plain

On Thu, 11 Sep 1997, Alan Bloom wrote:

> I don't see how a class C transmitter will work with QAM (or DSB AM
> for that matter). You will likely get huge symbol errors as well as
> horrible adjacent-channel interference.

Hi Alan,

I imagine one way might be to repartition the decision regions for each symbol on the receive end to account for the phase and magnitude non-linearities. I have seen this done for QAM transmissions through the TDRS satellite where the TWT is run in saturation.

For more information, I can probably point you to a PhD dissertation at NMSU a few years ago.

73,
Tim AA5DF

33. Sorry for EVM-Memory-Extens.

From: "Ulrich G. Schaarschmidt" <ulrich.schaarschmidt@UniBw-Hamburg.DE>
Date: Mon, 22 Sep 1997 09:38:24 +0000
To: hfsig@tapr.org
Message-ID: <199709220739.JAA19900@rzsparc-3.unibw-hamburg.de>
Rendered body type: text/plain

Dear group,
I am sorry for the fat mailing.
I had a lot of boys, requesting the pages and for one
answer I forgot to change the back-address.

Please excuse my mistake.
73 de Uli, DJ1HM

34. Re: [HFSIG:2463] Sorry for EVM-Memory-Extens.

From: ADIDAS753@aol.com
Date: Mon, 22 Sep 1997 06:14:04 -0400 (EDT)
To: hfsig@tapr.org

Message-ID: <970922061403_-1565369228@emout07.mail.aol.com>
Rendered body type: text/plain

someone sign me off this thing

35. Re: [HFSIG:2464] Re: Sorry for EVM-Memory-Extens.

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 22 Sep 1997 08:20:36 -0800
To: hfsig@tapr.org
Message-ID: <199709221521.IAA02551@peak.org>
Rendered body type: text/plain

>someone sign me off this thing
>
>

Nobody can "sign you off" - if that was possible, then people can remove
others as they wish!

You need to point your web browser to www.tapr.org. Go down to the special
interest groups. There is a box that you can check to remeove you from
selected special interest groups. Check HFSIG and submit your request.

If you do this, then the web server will verify addresses against the one
trying to remove you and if they are the same, you will be removed.

36. WTB: DSP-93

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 22 Sep 1997 08:20:38 -0800
To: DSP@tapr.org, HFSIG@tapr.org
Message-ID: <199709221522.IAA02559@peak.org>
Rendered body type: text/plain

Hi,

I require a DSP-93 (or two) for a project and would be interested in
purchasing a used DSP-93 or unassembled kit.

If anyone considers selling his DSP-93 that is in perfect working order, or
still in unassembled form, please e-mail me directly and state condition and
price. I would prefer offers from within the USA in order to avoid shipping
and customs issues.

Thank you much in advance for your help.

--Johan Forrer
(forrerj@peak.org)

37. Re: [HFSIG:2462] Re: digital test

From: Antony Martin <anthony.martin@gecm.com>
Date: Mon, 22 Sep 1997 17:04:27 +0100
To: hfsig@tapr.org
Message-ID: <970922170427.8630@oas01.gmcl.co.uk>
Rendered body type: text/plain

> On Thu, 11 Sep 1997, Alan Bloom wrote:
>
> > I don't see how a class C transmitter will work with QAM (or DSB AM
> > for that matter). You will likely get huge symbol errors as well as
> > horrible adjacent-channel interference.
>
> Hi Alan,
>
> I imagine one way might be to repartition the decision regions for each
> symbol on the receive end to account for the phase and magnitude
> non-linearities. I have seen this done for QAM transmissions through the
> TDRS satellite where the TWT is run in saturation.
>
> For more information, I can probably point you to a PhD dissertation at
> NMSU a few years ago.
>
> 73,
> Tim AA5DF
>

You would have to adjust levels at the Tx end for this to be possible,
otherwise all transmitted amplitude levels would be virtually the same.

SSB HF transceivers have tight control of gain and linearity of all

stages and include ALC systems etc.

The problem with class C (ie. broadcast AM) HF transmitters is that no such control is required. The driver stages may also be class C and saturate. The gain may vary considerably over the multioctave BW and also with temperature, supply etc. The non-linearities will therefore not be stable. Also, operating Class C power devices at lower drive to deliberately bring them into the linear region will increase their internal dissipation and cause failure.

If you want to transmit QAM, DSB, ISB, SSB etc. from a class-C AM broadcast transmitter you need to use EER techniques. We've recently done an EER variant of our digital transmitter drive for customers that want the capability for SSB on their 500kW AM transmitters.

As JPL are the people who put the Pathfinder on Mars, JPL aren't stupid; they're probably already doing this. If not, buy one of our EER Drives.

I don't quite see the sense of measuring things like adjacent channel interference on-air; isn't it better to do it off-air with a spectrum analyser or HF receiver? But I suppose a dummy load for serious AM broadcast powers (500kW) is a non-trivial installation and I doubt many exist worldwide.

Unfortunately I missed the test transmission as I was on holiday. It would have been nice to monitor & sample the signals.

Footnote: EER (Envelope elimination & restoration)
- --------
This is when the waveform to be transmitted is split into phase and amplitude components. The phase component as a constant-amplitude RF signal drives the class-C or switching output stage. The amplitude component is applied to the supply of the output stage. This is often done by some form of switching regulator. The signals, when combined with necessary compensating delays, form the wanted output waveform.

EER results in high efficiency. Some would say at the expense of complexity. But when there's megawatts at stake, it gives large savings on the cost of electricity, the size of the cooling plant, the life of the valve...

Anthony

_____

A.N.Martin   E-mail anthony.martin@gecm.com
Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
_____

38. TMS320C50 version for PSK31

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 22 Sep 1997 09:46:04 -0800
To: DSP@tapr.org, HFSIG@tapr.org
Message-ID: <199709221647.JAA16577@peak.org>
Rendered body type: text/plain

Hi,

I have uploaded Andrew's, G0TJZ, version of PSK31 for the TI320C50 DSK to the DSP file area. It will be available in the "recent_uploads" directory as soon as Tim has moved it.

Thanks to Andrew for this contribution. This came via Paul, PA0OCD. Thanks Paul.

73's

--Johan, KC7WW

39. Re: [HFSIG:2460] RE: Extended memory for 56002EVM

From: David A Willmore <willmore@cig.mot.com>
Date: Mon, 22 Sep 1997 13:27:56 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <9709221327.ZM21606@whelk>
Rendered body type: text/plain

On Sep 19, 12:15pm, David A Willmore wrote:
> Ok, this is my second take at your message. I spent some of last night

staring
> at this map and you're right, I was confised when I wrote it. Here's the
> corrected version:
>
>   PROG   X   Y
>   +------+------+-----+
> 7FFF | F | I/O | |
> | L +------+ S |
> ... | A | | R |
> | S | | A |
> 8000 | H | | M |
>   +------+ | |
> 7FFF | | Y |
> ... | SRAM X/P | |
> 200 | | |
>   +------+------+-----+
> 1FF ... INTERNAL
> 000
>
> This looks like it's going to be easy to decode--I'm crunching the K-maps in
my
> spare time.
>
> More when it's available....

It's available...

To implement the above memory map, it would take five NAND gates and would be
two gates deep. That's quite a bit of delay for something with a short time to
play with. The other option is to remove the shared SRAM between X and P in
the 200-7FFF region. I can do that--preserving the FLASH in P space--with no
gates.

I have yet to run a timing analysis on the five gate solution, it might not be
a problem. Is it important to have that memory shared between X and P? I would
assume so, but I haven't written code for this thing. Opinions? (email me
personally and I'll summarize)

Cheers,
David

40. Re: [HFSIG:2467] Re: digital test

From: Arvydas Vaisnys <vaisnys@zorba.jpl.nasa.gov>
Date: Mon, 22 Sep 1997 15:26:04 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19970922152602.006a63d0@zorba.jpl.nasa.gov>
Rendered body type: text/plain

At 11:10 AM 9/22/97 -0500, you wrote:
>The problem with class C (ie. broadcast AM) HF transmitters is that no
>such control is required. The driver stages may also be class C and
>saturate. The gain may vary considerably over the multioctave BW and
>also with temperature, supply etc. The non-linearities will therefore
>not be stable. Also, operating Class C power devices at lower drive to
>deliberately bring them into the linear region will increase their
>internal dissipation and cause failure.
>
>If you want to transmit QAM, DSB, ISB, SSB etc. from a class-C AM
>broadcast transmitter you need to use EER techniques. We've recently
>done an EER variant of our digital transmitter drive for customers that
>want the capability for SSB on their 500kW AM transmitters.
>
>As JPL are the people who put the Pathfinder on Mars, JPL aren't stupid;
>they're probably already doing this. If not, buy one of our EER Drives.
>
>I don't quite see the sense of measuring things like adjacent channel
>interference on-air; isn't it better to do it off-air with a spectrum
>analyser or HF receiver? But I suppose a dummy load for serious AM
>broadcast powers (500kW) is a non-trivial installation and I doubt
>many exist worldwide.
>
>Unfortunately I missed the test transmission as I was on holiday. It
>would have been nice to monitor & sample the signals.
>
>Footnote: EER (Envelope elimination & restoration)
>- --------
>This is when the waveform to be transmitted is split into phase and
>amplitude components. The phase component as a constant-amplitude
>RF signal drives the class-C or switching output stage. The amplitude

>component is applied to the supply of the output stage. This is often
>done by some form of switching regulator. The signals, when combined
>with necessary compensating delays, form the wanted output waveform.
>
>EER results in high efficiency. Some would say at the expense of
>complexity. But when there's megawatts at stake, it gives large
>savings on the cost of electricity, the size of the cooling plant,
>the life of the valve...
>
>Anthony
>
>____
>
>A.N.Martin   E-mail anthony.martin@gecm.com
>Development Engineer   Tel.   01245 353221 x4995
>Military Products Engineering   Fax.   01245 275993
>GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
>____

Thanks for the confidence in JPL and sorry for my sloppy usage of amplifier terminology. What we used in the test was a fairly standard, high power (250 kW), plate modulated DSB AM transmitter. I don't know if the amplification is Class C, but it certainly does not like to amplify signals with amplitude components, such as pulse shaped MPSK or QAM, when these signals are put into the exciter input.

Yes we did use what you call an EER technique, perhaps in a slightly simpler manner: The signal is split into a constant amplitude phase modulated carrier and an amplitude component. The phase component goes into the exciter, while the amplitude component is applied to the normal AM modulation input. Since the AM modulation input is AC coupled and has a finite frequency response, the signal does not get perfectly restored. Nevertheless, with fairly simple delay and gain adjustments, we were able to get the near sidelobes at least 30 dB below the main lobe.

Also apologies to those trying to listen. We got started about an hour late (by 17:00 UT) on 15.24 Mhz and ran about an hour each at 220 kW and 50 kW. At 19:00 we found that there was a strong broadcast in French on 11.975 MHz so we had to switch to 12.03 MHz and transmitted for two more hours (not that this frequency or 15.24 MHz was interference free either).

In both this test an a previous test in May, we found the Delano to Washington DC propagation path to be very difficult. Perhaps this has to do with Washington being the receive site. In May it looked like a classical three component Rayleigh channel. This time there was slightly less multipath, but there was a strong AM signal 5kHz away.

Nevertheless, with suitable error correction coding and equalization, we have managed to get error free compressed (16 kbps) audio at 8PSK at an 8k symbol rate. Analysis is still going on to see how well things work at higher MPSK and QAM. Our receiver is a normal short wave receiver with an extra downconversion to an IF at 13 KHz. This IF is recorded on a DAT at a 48 kHz sampling rate. Demodulation and the rest of the processing is done in non-real time on a PC.

One interesting result is that the receiver reaches an error floor when propagation is over a multipath channel. Thus increasing transmitted power does not help after a certain point. Error correction coding must be used to bring the bit error rate down from this floor to a level that is acceptable to the audio decoder. (The higher the modulation M, the higher this error floor ends up at).

As far as far as dummy loads are concerned, Delano has a dummy load that can handle 250 kW, and we used it to look at spectra. However tolerable interference ratios will only be established through listening tests. In most broadcasting systems it's what the listener will tolerate that counts.

Arv

Arvydas Vaisnys
MS 161-260
Jet Propulsion Laboratory
4800 Oak Grove Drive
Pasadena, CA 91109
Tel: 818-354-6219
Fax: 818-393-4643

41. RE: [HFSIG:2469] RE: Extended memory for 56002EVM

From: jalocha@axdele.cern.ch
Date: Tue, 23 Sep 1997 11:30:21 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199709230930.LAA32307@dxmint.cern.ch>
Rendered body type: text/plain

>I have yet to run a timing analysis on the five gate solution, it might not be
>a problem. Is it important to have that memory shared between X and P? I would
>assume so, but I haven't written code for this thing. Opinions? (email me
>personally and I'll summarize)

My view: I can't see any practical use for shared memory space
(so that same physical bytes of a RAM are seen at different addresses
or memory spaces). I always thought the shared space is a not-really-desired
byproduct of a simplified/zero-glue-logic RAM decoding.
My software doesn't use such features at all. The DSP56002 can address
all X/Y/P spaces without tricks.

If you make decoding the way that all X/Y/P spaces have their proper
physical locations I would be happy :-)

One thing I care for: I like lot of L space, where I have X and Y space
filled so I can store long (48-bit) or complex numbers.
This is usefull for storing FFT results, vector products,
correlation/multiplication outputs, etc. Ideally one should fill
P:8000-FFFF with EEPROM and all the rest with RAM with no sharing anywhere.

Pawel

42. DSP-93

From: Mel Roberts <RobertsM@jsc.mil>
Date: Tue, 23 Sep 1997 07:21:15 -0400
To: hfsig@tapr.org
Message-ID: <s4276e0b.003@jsc.mil>
Rendered body type: text/plain

Johan,
  Have an assembled, working DSP-93 for sale for $400 including shipping.
Includes all documentation and power supply.
  73 Mel Roberts W3MR robertsm@jsc.mil or w3mr@erols.com

43. Re: [HFSIG:2471] RE: Extended memory for 56002EVM

From: David A Willmore <willmore@cig.mot.com>
Date: Tue, 23 Sep 1997 10:38:32 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <9709231038.ZM10123@whelk>
Rendered body type: text/plain

On Sep 23, 4:37am, jalocha@axdele.cern.ch wrote:
> My view: I can't see any practical use for shared memory space
> (so that same physical bytes of a RAM are seen at different addresses
> or memory spaces). I always thought the shared space is a not-really-desired
> byproduct of a simplified/zero-glue-logic RAM decoding.
> My software doesn't use such features at all. The DSP56002 can address
> all X/Y/P spaces without tricks.
>
> If you make decoding the way that all X/Y/P spaces have their proper
> physical locations I would be happy :-)
>
> One thing I care for: I like lot of L space, where I have X and Y space
> filled so I can store long (48-bit) or complex numbers.
> This is usefull for storing FFT results, vector products,
> correlation/multiplication outputs, etc. Ideally one should fill
> P:8000-FFFF with EEPROM and all the rest with RAM with no sharing anywhere.

Splendid. The easiest scheme seems to be to map X and Y seperately and to
leave P empty. I don't think that will make too many people happy as that only
leaves the on chip P RAM. I imagine that will cause people a little stress.
 To leave the onboard SRAM as P memory, it would take cutting a trace or two
and tacking on a gate. ~CE for the SRAM needs to be ~PS OR A15, not just A15.

I still haven't looked at the timing budget, but I expect the X and Y SRAM will
be zero wait state, but that the P might be one. Would that kill anyone?

Cheers,
David

44. Re: WTB: DSP-93

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 23 Sep 1997 10:21:25 -0800
To: HFSIG@tapr.org, DSP@tapr.org
Message-ID: <199709231722.KAA25342@peak.org>
Rendered body type: text/plain

Hi,

Thank you all for the response for help that I recieved - I now have all the
units that I need.

73's

--Johan

45. Re: [HFSIG:2473] RE: Extended memory for 56002EVM

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 23 Sep 1997 10:33:31 -0800
To: hfsig@tapr.org
Message-ID: <199709231734.KAA27113@peak.org>
Rendered body type: text/plain

David,

>Splendid. The easiest scheme seems to be to map X and Y seperately and to
>leave P empty. I don't think that will make too many people happy as that only
>leaves the on chip P RAM. I imagine that will cause people a little stress.
> To leave the onboard SRAM as P memory, it would take cutting a trace or two
>and tacking on a gate. ~CE for the SRAM needs to be ~PS OR A15, not just A15.
>
>I still haven't looked at the timing budget, but I expect the X and Y SRAM will
>be zero wait state, but that the P might be one. Would that kill anyone?

Yes, unfortunately it will impact the whole body of existing code - and
there is quite a bit of that. From other designs that I have seen, you can
do it without any extra gates. The DSP SRAM timing is really very tight.

Good luck.

--Johan

46. Re: [HFSIG:2467] Re: digital test

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Tue, 23 Sep 1997 12:27:35 -0600 (MDT)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.3.96.970923114002.2576C-100000@murdrum>
Rendered body type: text/plain

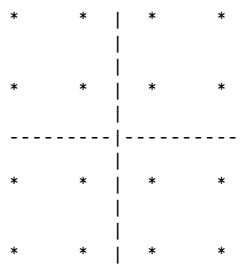
On Mon, 22 Sep 1997, Antony Martin wrote:

> > I imagine one way might be to repartition the decision regions for each
> > symbol on the receive end to account for the phase and magnitude
> > non-linearities. I have seen this done for QAM transmissions through the
> > TDRS satellite where the TWT is run in saturation.
> >
> > 73,
> > Tim AA5DF
> >
>
> You would have to adjust levels at the Tx end for this to be possible,
> otherwise all transmitted amplitude levels would be virtually the same.
>

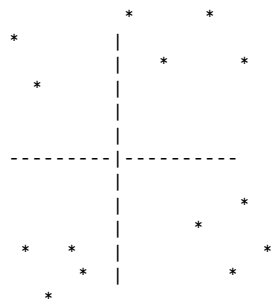
Hi Anthony,

Yes, you are correct. The method I mentioned is an attempr around this
limitation. I am not all that familiar with satellite TWT amps, I believe
they are class C. Due to the use of TWT amps in satellites is typically
the reason PSK is used on satellite channels. When you run a non-constant
amplitude signal through the TWT, such as QAM, the smaller magnitude
components may be amplified, say, 15. Yet the components with a greater
magnitude will still be amplitfied, they may only get amplified to a
lesser amount, perhaps only 110%. The greater magnitude components will
also exhibit a greater phase shift.

Let me attempt a constellation diagram:



Constelation before the TWT amplifier



Constallation after the TWT amplifier.

Okay, so the last constellation may not be perfect. It is a real challenge
to draw it in ASCII. Allow your imagination to fill in the gaps.

If you know the characterisitcs of the amplifier such as it's phase shift
and amplitude variation for a spcific input, then you can repartition your
receiver's decision regions based on where the received symbols should be.
I'm sure this would not give you ideal performance compared to a linear
QAM system, but it was simulated and determined the idea would be
workable.

Would this work in amateur radio? probably not. Interesting idea though.

If anyone wants more information, email me privately, and I'll try to
direct you to the disseration.

The EER technique you describe is quite interesting as well. Thanks for
sharing.

73,
Tim Baggett

AA5DF
tim@acca.nmsu.edu

47. Re[2]: [HFSIG:2467] Re: digital test

From: "Marion Hales" <Marion_Hales@BENG.VOA.GOV>
Date: Tue, 23 Sep 97 16:18:54 EST
To: hfsig@tapr.org, Arvydas Vaisnys <vaisnys@zorba.jpl.nasa.gov>
Cc: George_Woodard@BENG.VOA.GOV, Vince_Nowicki@BENG.VOA.GOV,
Bill_May@BENG.VOA.GOV, Robert_Everett@BENG.VOA.GOV
Message-ID: <9708238750.AA875057186@ccmail.beng.voa.gov>
Rendered body type: text/plain

Arv,

Thank you for forwarding this to me. I believe your response to Mr.
Martin was quite correct.

Mr. Martin refers to a Marconi exciter that might be adaptable to your
modulation scheme. I'm not familiar with it, but I will be in
Chelmsford next week to visit the Marconi plant so I'll try to get
info as it could be used to upgrade the B6128 transmitters at some

point in the future. Continental in Dallas also makes an exciter that may be adaptable to your method and to the B6128. I'll try to get info on that one as well.

Sounds like you have decoded some data. How does it look?

Regards,

Marion

_____ Reply Separator _____
Subject: Re: [HFSIG:2467] Re: digital test
Author: Arvydas Vaisnys <vaisnys@zorba.jpl.nasa.gov> at Internet-Mail
Date: 9/22/97 6:46 PM

At 11:10 AM 9/22/97 -0500, you wrote:
>The problem with class C (ie. broadcast AM) HF transmitters is that no
>such control is required. The driver stages may also be class C and
>saturate. The gain may vary considerably over the multioctave BW and
>also with temperature, supply etc. The non-linearities will therefore
>not be stable. Also, operating Class C power devices at lower drive to
>deliberately bring them into the linear region will increase their
>internal dissipation and cause failure.
>
>If you want to transmit QAM, DSB, ISB, SSB etc. from a class-C AM
>broadcast transmitter you need to use EER techniques. We've recently
>done an EER variant of our digital transmitter drive for customers that
>want the capability for SSB on their 500kW AM transmitters.
>
>As JPL are the people who put the Pathfinder on Mars, JPL aren't stupid;
>they're probably already doing this. If not, buy one of our EER Drives.
>
>I don't quite see the sense of measuring things like adjacent channel
>interference on-air; isn't it better to do it off-air with a spectrum
>analyser or HF receiver? But I suppose a dummy load for serious AM
>broadcast powers (500kW) is a non-trivial installation and I doubt
>many exist worldwide.
>
>Unfortunately I missed the test transmission as I was on holiday. It
>would have been nice to monitor & sample the signals.
>
>Footnote: EER (Envelope elimination & restoration)
>- --------
>This is when the waveform to be transmitted is split into phase and
>amplitude components. The phase component as a constant-amplitude
>RF signal drives the class-C or switching output stage. The amplitude
>component is applied to the supply of the output stage. This is often
>done by some form of switching regulator. The signals, when combined
>with necessary compensating delays, form the wanted output waveform.
>
>EER results in high efficiency. Some would say at the expense of
>complexity. But when there's megawatts at stake, it gives large
>savings on the cost of electricity, the size of the cooling plant,
>the life of the valve...
>
>Anthony
>
>_____
>
>A.N.Martin   E-mail anthony.martin@gecm.com
>Development Engineer   Tel.   01245 353221 x4995
>Military Products Engineering   Fax.   01245 275993
>GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
>_____

Thanks for the confidence in JPL and sorry for my sloppy usage of amplifier terminology. What we used in the test was a fairly standard, high power (250 kW), plate modulated DSB AM transmitter. I don't know if the amplification is Class C, but it certainly does not like to amplify signals with amplitude components, such as pulse shaped MPSK or QAM, when these signals are put into the exciter input.

Yes we did use what you call an EER technique, perhaps in a slightly simpler manner: The signal is split into a constant amplitude phase modulated carrier and an amplitude component. The phase component goes into the exciter, while the amplitude component is applied to the normal AM

modulation input. Since the AM modulation input is AC coupled and has a finite frequency response, the signal does not get perfectly restored. Nevertheless, with fairly simple delay and gain adjustments, we were able to get the near sidelobes at least 30 dB below the main lobe.

Also apologies to those trying to listen. We got started about an hour late (by 17:00 UT) on 15.24 Mhz and ran about an hour each at 220 kW and 50 kW. At 19:00 we found that there was a strong broadcast in French on 11.975 MHz so we had to switch to 12.03 MHz and transmitted for two more hours (not that this frequency or 15.24 MHz was interference free either).

In both this test an a previous test in May, we found the Delano to Washington DC propagation path to be very difficult. Perhaps this has to do with Washington being the receive site. In May it looked like a classical three component Rayleigh channel. This time there was slightly less multipath, but there was a strong AM signal 5kHz away.

Nevertheless, with suitable error correction coding and equalization, we have managed to get error free compressed (16 kbps) audio at 8PSK at an 8k symbol rate. Analysis is still going on to see how well things work at higher MPSK and QAM. Our receiver is a normal short wave receiver with an extra downconversion to an IF at 13 KHz. This IF is recorded on a DAT at a 48 kHz sampling rate. Demodulation and the rest of the processing is done in non-real time on a PC.

One interesting result is that the receiver reaches an error floor when propagation is over a multipath channel. Thus increasing transmitted power does not help after a certain point. Error correction coding must be used to bring the bit error rate down from this floor to a level that is acceptable to the audio decoder. (The higher the modulation M, the higher this error floor ends up at).

As far as far as dummy loads are concerned, Delano has a dummy load that can handle 250 kW, and we used it to look at spectra. However tolerable interference ratios will only be established through listening tests. In most broadcasting systems it's what the listener will tolerate that counts.

Arv

Arvydas Vaisnys
MS 161-260
Jet Propulsion Laboratory
4800 Oak Grove Drive
Pasadena, CA 91109
Tel: 818-354-6219
Fax: 818-393-4643

48. Re: [HFSIG:2470] Re: digital test

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 23 Sep 1997 17:48:44 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <m0xDfdE-000HRQC@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>One interesting result is that the receiver reaches an error floor when
>propagation is over a multipath channel. Thus increasing transmitted power
>does not help after a certain point. Error correction coding must be used
>to bring the bit error rate down from this floor to a level that is
>acceptable to the audio decoder. (The higher the modulation M, the higher
>this error floor ends up at).

Not just coding, but also interleaving (particularly with short block or convolutional codes). You must spread the coded symbols out over a much greater interval than the channel fade rate. When fading is slow, this can add significant delay. But it may be acceptable for one-way broadcasting.

Phil

49. Re: [HFSIG:2462] Re: digital test

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 23 Sep 1997 17:25:42 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <m0xDfGw-000HRMC@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>I imagine one way might be to repartition the decision regions for each
>symbol on the receive end to account for the phase and magnitude
>non-linearities. I have seen this done for QAM transmissions through the

>TDRS satellite where the TWT is run in saturation.

TWTs may be nonlinear when driven at high levels, but they are *not*
running in Class C. They're quite linear at low levels, and you can
still compensate (at least partially) for the high power
nonlinearities.

UHF TV transmitter klystrons work the same way. When I engineered for
WMPB-TV (ch 67) in Baltimore in the 1970s, we used a nonlinear
processing amplifier to stretch the video sync pulse to overcome the
klystron's inherent nonlinearity at high power levels.

The US TV VSB-AM broadcast standard specifies inverted video. Minimum
power is white, and greater power levels correspond to darker
luminance levels. (White doesn't actually go to zero power, as a
certain minimum video power is required at all times to correctly
demodulate the sound subcarrier in most TV sets. Overmodulation, e.g.,
on white text overlays, causes a characteristic 60Hz audio buzz in
these TVs.)

Sync is "blacker than black", so it corresponds to the peak
transmitter output power. Why inversion was specified should now be
obvious -- inverting the video signal keeps the fragile analog video
information within the linear range of the klystron. The sync pulse,
being 'digital-like', can easily tolerate the nonlinearities of the
processing amp and transmitter.

(I always wanted to try a test to see if the klystron temperature
and/or cooling water boiloff rate depended noticeably on scene
brightness. In theory, it should. These tubes draw a constant DC
input regardless of drive level -- 18 kV @ 4.5A for each of the two
visual tubes plus 2.25A for the single aural tube -- so a bright scene
results in less of the DC input being converted into RF and more of it
into heat. Conversely, a black scene should result in a cooler
transmitter.)

A Class C amplifier allows no envelope variations at all, unless you
impose them by some form of plate or collector modulation as in an AM
transmitter or a EER (Envelope Elimination and Restoration)
scheme. The most clever such scheme I've seen was Karl Meinzer's
HELAPS (High Efficiency Linear Amplification through Parametric
Synthesis) for AO-13. In HELAPS you have two parallel final stages.
Both finals operate in Class C, and the outputs are combined before
reaching the antenna. The inputs, however, come from a
voltage-controlled phase splitter. The control voltage comes from an
amplitude detector on the drive signal. When the desired output
amplitude is at a minimum, the two finals are driven out of phase so
their outputs also cancel, producing a minimum signal at the
antenna. When max power is desired, the two finals are driven in phase
so their outputs add.

Obviously this takes very careful design of the final stages to keep
them stable, as they're effectively operating into a rapidly varying
load impedance. But the payoff is worth it; the result was one of the
most efficient linear power amplifiers ever built.

Phil

50. Re: [HFSIG:2479] Re: digital test

From: Lynn Coffelt <lynnc@sos.net>
Date: Wed, 24 Sep 1997 05:35:30 -0700
To: hfsig@tapr.org
Message-ID: <34290911.D1E156C7@sos.net>
Rendered body type: text/plain

Phil, certainly enjoyed lurking into the AM modulation explanations. In the
late '40s, we hams were constantly striving to beat the high cost of plate
modulation transformers by experimenting with various control grid and screen
grid modulation techniques. Keep up the great
work!   Lynn, W7LTQ

Phil Karn wrote:

> >I imagine one way might be to repartition the decision regions for each
> >symbol on the receive end to account for the phase and magnitude
> >non-linearities. I have seen this done for QAM transmissions through the
> >TDRS satellite where the TWT is run in saturation.
>
> TWTs may be nonlinear when driven at high levels, but they are *not*

> running in Class C. They're quite linear at low levels, and you can
> still compensate (at least partially) for the high power
> nonlinearities.
>
> UHF TV transmitter klystrons work the same way. When I engineered for
> WMPB-TV (ch 67) in Baltimore in the 1970s, we used a nonlinear
> processing amplifier to stretch the video sync pulse to overcome the
> klystron's inherent nonlinearity at high power levels.
>
> The US TV VSB-AM broadcast standard specifies inverted video. Minimum
> power is white, and greater power levels correspond to darker
> luminance levels. (White doesn't actually go to zero power, as a
> certain minimum video power is required at all times to correctly
> demodulate the sound subcarrier in most TV sets. Overmodulation, e.g.,
> on white text overlays, causes a characteristic 60Hz audio buzz in
> these TVs.)
>
> Sync is "blacker than black", so it corresponds to the peak
> transmitter output power. Why inversion was specified should now be
> obvious -- inverting the video signal keeps the fragile analog video
> information within the linear range of the klystron. The sync pulse,
> being 'digital-like', can easily tolerate the nonlinearities of the
> processing amp and transmitter.
>
> (I always wanted to try a test to see if the klystron temperature
> and/or cooling water boiloff rate depended noticeably on scene
> brightness. In theory, it should. These tubes draw a constant DC
> input regardless of drive level -- 18 kV @ 4.5A for each of the two
> visual tubes plus 2.25A for the single aural tube -- so a bright scene
> results in less of the DC input being converted into RF and more of it
> into heat. Conversely, a black scene should result in a cooler
> transmitter.)
>
> A Class C amplifier allows no envelope variations at all, unless you
> impose them by some form of plate or collector modulation as in an AM
> transmitter or a EER (Envelope Elimination and Restoration)
> scheme. The most clever such scheme I've seen was Karl Meinzer's
> HELAPS (High Efficiency Linear Amplification through Parametric
> Synthesis) for AO-13. In HELAPS you have two parallel final stages.
> Both finals operate in Class C, and the outputs are combined before
> reaching the antenna. The inputs, however, come from a
> voltage-controlled phase splitter. The control voltage comes from an
> amplitude detector on the drive signal. When the desired output
> amplitude is at a minimum, the two finals are driven out of phase so
> their outputs also cancel, producing a minimum signal at the
> antenna. When max power is desired, the two finals are driven in phase
> so their outputs add.
>
> Obviously this takes very careful design of the final stages to keep
> them stable, as they're effectively operating into a rapidly varying
> load impedance. But the payoff is worth it; the result was one of the
> most efficient linear power amplifiers ever built.
>
> Phil

51. Re: [HFSIG:2479] digital test

From: Tom McDermott <mcdermot@aud.alcatel.com>
Date: Wed, 24 Sep 1997 08:08:22 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BCC8C1.07446F20.mcdermot@aud.alcatel.com>
Rendered body type: text/plain

>I imagine one way might be to repartition the decision regions for each
>symbol on the receive end to account for the phase and magnitude
>non-linearities. I have seen this done for QAM transmissions through the
>TDRS satellite where the TWT is run in saturation.

In older digital microwave transmitters, the TWT is in fact fairly linear. (not class C). In order to achieve acceptable performance for QAM transmission, the TWT is operated substantially below it's rated capability. Typically, a 6 Ghz. commercial transmitter will produce 5 watts output, but a 50-watt TWT used - it is operated class A and thus produces a lot of waste heat. In order to improve the efficiency of TWT's, a lot of work was done in the 1980's on "pre-distortion" equipment.

Pre-distortion consists mainly of passing the drive signal through a non-linear amplitude function that compensates for the TWT gain-compression at higher power levels. I think that there are also some unusual phase-change effects at the high power levels, but I don't really recall.

Current GaAs-FET based power ampliifers in the commercial microwave bands must also be
operated substantially below the rated power level due to non-linearity of the FET.

Tom McDermott
mcdermot@aud.alcatel.com

52. Re: [DSP:680] Re: MMX-aware assembler

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Tue, 30 Sep 1997 19:28:49 +0200
To: Johan Forrer <forrerj@peak.org>
Cc: hfsig@tapr.org
Message-ID: <343136D1.3CA@ife.ee.ethz.ch>
Rendered body type: text/plain

> Could you give a specific URL. I have heard from others that that is what

ftp://prep.ai.mit.edu is the main site, I'm sure they list their mirrors...

> be at an experimental level. I dont think it is available at _all_ mirrors.

No, it's quite an official release, I doubt that any GNU mirror misses
it...

> It would be nice to use it with DJGPP, which I believe will work with
> binutils 2.8.1.

Alternatively, you could try to compile it with the cygwin kit
to run it under windows 32bit, see ftp://ftp.cygnus.com

However, the compile scripts of the GNU utils are _way_ complex,
don't know how this works under non-Unix...

73s
Tom

53. Re: [HFSIG:2482] Re: [DSP:680] Re: MMX-aware assembler

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 30 Sep 1997 20:51:04 +0100
To: hfsig@tapr.org
Message-ID: <34315828.AE4CFFAB@dircon.co.uk>
Rendered body type: text/plain

Thomas Sailer wrote:

> However, the compile scripts of the GNU utils are _way_ complex,
> don't know how this works under non-Unix...
>
> 73s
> Tom

It can be done but you need to get just about every DOS implementationof the UNIX environment to do it.
Then you get problems with
file name lengths
and files that have multiple dots in it etc.

- Charles G4GUO

54. 1997 ARRL and TAPR DCC Reminder

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Tue, 30 Sep 1997 15:15:53 -0500
To: " TAPR/AMSAT DSP "<dsp@tapr.org>, " Spread Spectrum "<ss@tapr.org>,
"HF SIG list mailing"<hfsig@tapr.org>,
"NETSIG list mailing"<netsig@tapr.org>,
"TAPR APRS SIG"<aprssig@tapr.org>, " MIC-E Beta "<mic-e@tapr.org>,
" tacgps "<tacgps@tapr.org>
Message-ID: <v03110717b0570c981bd4@[208.134.134.40]>
Rendered body type: text/plain

Don't forget that the 16th Annual ARRL and TAPR Digital Communications
Conference is in just a few days!

It is not to late to make the conference. Registration is already at a
high and we expect a really great conference! Hope to see you in Baltimore.

1997 ARRL and TAPR Digital Communications Conference
October 10-12, 1997
Baltimore, Maryland (minutes from BWI airport)

Main Web Page -- http://www.tapr.org/dcc
Tentative Schedule -- http://www.tapr.org/tapr/html/dcc97schedule.html
Proceedings Abstracts -- http://www.tapr.org/tapr/html/cnc16.html
Local Attractions -- http://www.tapr.org/tapr/html/dcc97_attractions.html

---

The 1997 ARRL and TAPR Digital Communications Conference will be held on October 10-12, 1997 in Baltimore, Maryland. This years conference location is just minutes away from the BWI Airport.

What can you expect during the 1997 ARRL and TAPR Digital Communications Conference ?
* A full day of papers and breakouts on Saturday for the beginner to the advanced amateur digital enthusiast.

* Three workshops
  - o Friday (1pm - 8pm) - APRS Symposium (http://www.aprs.net/dcc/)
  - o Friday (4pm - 8pm) - Seminar on RF Basics for Computer Weenies
  - o Sunday (8:30am - 1:30pm) - Seminar on Spread Spectrum system Design and Theory

* The second annual Student paper session.

* A banquet with a Special Guest Speaker Yutaka Sakurai, JF1LZQ who is Vice President of Japan's Packet Radio User's Group (PRUG). His talk is entitled, "Japan's PRUG: A look at its history and a view to the future". He will discuss the history of the PRUG from its early beginnings in the mid-80's and track its development to today. He will also discuss some of PRUG's current projects and its goals for the future.

* SIGs (Special Interest Groups) on Saturday following the banquet.

* Informal get-togethers throughout the weekend.

* TAPR Membership Meeting

Special Symposiums/Seminars

In addition to the presentation of papers on Saturday, three symposiums/seminars will be held before and after the conference. These sessions are provided to allow those with additional time and interest to make the most of the Digital Communications Conference. For those that might have interest in just a symposium or seminar, registration for the conference is not necessary to register and attend one or more of the symposiums and seminars. This allows maximum flexibility for those that might want to participate during the Digital Communications Conference, but do not have an entire weekend to devote to the event.

A full day symposium on Friday covering APRS will be conducted by Bob Bruninga, WB4APR (father of APRS) Keith Sproul, WU2Z, Mark Sproul, KB2ICI (developers of Mac and Windows APRS), Steve Dimse, K4HG (developer of javAPRS), and other nationally known APRS leaders. Join this group for the afternoon and evening for in depth discussions and presentations on the current and future status of APRS. This is an unique opportunity to gain insight into this fast growing digital aspect of amateur operations that combines computers, packet radio, and GPS (Global Positioning System). See http://www.aprs.net/dcc/ for schedule.

Starting late Friday afternoon a half-day seminar entitled "RF Basics for Computer Weenies: Helping the RF-challenged get the most out of the new high-speed wireless toys" presented by Mike Cheponis, K3MC. The seminar will focus on such topic as basic antenna types and their characteristics, comparison of feedlines and connector types, propagation basics (calculation of path loss, multipath effects, fresnel zones, etc), weatherproofing, lightning protection, filters, basic RF measurements, and other issues related to maintaining radio equipment on the UHF/SHF bands. This seminar will be starting late enough in the afternoon on Friday to allow those flying on that day to attend the seminar.

On Sunday morning, Dewayne Hendricks, WA8DZP, Phil Karn, KA9Q, and Tom McDermott, N5EG, will conduct a seminar focusing on "Spread Spectrum System Design and Theory." All three presenters are well known for their work in various areas of amateur digital communications and this 5 hour seminar should be an excellent opportunity to learn about the design and theory of spread spectrum systems. This seminar is a follow up to the past two seminars of PCS technology by Dewayne with a new focus on system design.

Airports

* The closest airport is Balitmore/Washington International (BWI)
* For those flying internationally or located on the East coast, check flights into
    - o Dulles International or
    - o Washington National Airport
* Bus/rail service from either Dulles or Washington to BWI is available, but be sure to check on this before arriving.

Hotel Information

Conference presentations, meetings, and workshops will be held at the Holiday Inn BWI Airport. DCC special rooms rates are $89 per night. When making reservations with the hotel, be sure to indicate you are attending the ARRL and TAPR DCC conference. The hotel provides transportation to and from BWI Airport, Amtrak/MARC rail station, and the local Baltimore Light Rail. You should contact the hotel to arrange transportation.

Holiday Inn BWI Airport (conference hotel)
890 Elkridge Landing Rd, Linthicum, MD, 21090
(410) 859-8400, Fax (410) 684-6778

3912.00N and 07641.34W

Directions to Conference

Coming from Baltimore Metro Area:
Follow MD-295 SOUTH towards BWI airport
Take W. NURSERY Rd exit
LEFT at the list at the end of the exit
RIGHT at the 3rd light, WINERSON Rd
LEFT at the 1st light, ELKRIDGE LANDING Rd
Holiday Inn BWI will be 1/2 mile on right.

Coming from Annapolis:
Follow I-97 NORTH/RT. 3 NORTH
Take DORSEY Rd Exit (rt 176 West)
RIGHT onto AVIATION Blvd (Rt 162)
Pass BWI Satellite parking
Follow signs to BWI Airport
RIGHT onto ELKRIDGE LANDING Rd
Holiday Inn BWI will be 1/2 mile on left.

Coming from Washington, DC:
Follow I-95 NORTH or MD-295 NORTH towards Baltimore
Take I-195 EAST towards BWI airport
Take EXIT 1A (Rt. 170 North)
LEFT at 2nd light, ELKRIDGE LANDING Rd
Holiday Inn BWI will be 1/2 mile on left.

Coming from the North:
Follow I-95 SOUTH or NEW JERSEY TURNPIKE SOUTH to I-95 South
Take I-95 SOUTH to I-195 EAST/AIRPORT to
Rt. 170 NORTH LINTHICUM to 2nd light,
turn left onto ELKRIDGE LANDING Rd
Holiday Inn BWI will be 1 mile on left.

Local Repeater Talk-In
Maryland FM Association WA3DZD repeater network has given premission to use their system for the talk-in on Saturday morning, October 11, 1997 and for use during the conference weekend. The frequecies are:

* 146.76 out, 146.16 in
* 224.76 out, 223.16 in
* 444.00 out, 449.00 in

The talk-in station will be using the call sign W3NSA.

Many thanks to the Maryland FM Association.

Tucson Amateur Packet Radio
8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399 * 940-383-0000
----
e-mail: tapr@tapr.org ftp: ftp.tapr.org   web: <http://www.tapr.org/>

October 1997

Original source file: hfsig.9710
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9710

1. Re: [HFSIG:2468] TMS320C50 version for PSK31

From: Antony Martin <anthony.martin@gecm.com>
Date: Wed, 01 Oct 1997 15:57:50 +0100
To: hfsig@tapr.org
Message-ID: <971001155750.19237@oas01.gmcl.co.uk>
Rendered body type: text/plain

Hi there.

Can anyone recommend a *good* 'terminal emulator' program for windows 95? I'm looking for a program that allows us to talk to RS232 devices at high data rates without dropping characters, and has upload and download facilities and a scripting language to automate things.

We've used 'Procomm' a lot. When windows 3.1 appeared we tried 'Terminal' but still we used 'Procomm' instead. Now we've got some PCs with windows 95 we've tried 'Hyperterminal', but we found it dropped characters uploading an intelhex file from a programmer, until we reduced the baud rate down to 9600. And you have to quit & restart the tool to get the baud rate to change, quite useless for scripting.

The tools are used to upload and download software to prom programmers and to boards which hold software in FLASH, testing modems, reading engineering & selftest data from modems, sending test commands etc.

I don't need bells and whistles for internet or dial-up use, just reliable access to the 232 ports at the higher rates newer PCs support.

Are you using a serial comms (terminal emulator) program that you would recommend? (shareware, freeware, PD or commercial) If so, please send me your suggestions.

Anthony.

ps. anyone seen any modem BER testing programs for win95???

A.N.Martin   E-mail anthony.martin@gecm.com
Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK

2. RE: [HFSIG:2485] Re: TMS320C50 version for PSK31

From: "Maples, David" <DMaples@sscpan.ssc.nasa.gov>
Date: Wed, 1 Oct 1997 12:04:30 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <EBC241916BECD01196840000F8045A680DB48C@sscpan.ssc.nasa.gov>
Rendered body type: text/plain

Folks around here use Procomm Rev. 4 for Windows 95 and seem to like it very well.

> -----Original Message-----
> From: Antony Martin [SMTP:anthony.martin@gecm.com]
> Sent: Wednesday, October 01, 1997 10:05 AM
> To: hfsig@tapr.org
> Subject: [HFSIG:2485] Re: TMS320C50 version for PSK31
>
> Hi there.
>
> Can anyone recommend a *good* 'terminal emulator' program for windows
> 95? I'm looking for a program that allows us to talk to RS232 devices
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> from a programmer, until we reduced the baud rate down to 9600.

> And you have to quit & restart the tool to get the baud rate
> to change, quite useless for scripting.
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> The tools are used to upload and download software to prom programmers
> and to boards which hold software in FLASH, testing modems, reading
> engineering & selftest data from modems, sending test commands etc.
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> I don't need bells and whistles for internet or dial-up use, just
> reliable access to the 232 ports at the higher rates newer PCs
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> Are you using a serial comms (terminal emulator) program that you
> would recommend? (shareware, freeware, PD or commercial) If so, please
> send me your suggestions.
>
> Anthony.
>
> ps. anyone seen any modem BER testing programs for win95???
>
> ____
> _
>
> A.N.Martin   E-mail anthony.martin@gecm.com
> Development Engineer   Tel.   01245 353221 x4995
> Military Products Engineering   Fax.   01245 275993
> GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
> ____
> _

3. How to measure Noise Figure of a receiver ?

From: Ted Manolescu <tmanolescu@instantel.com>
Date: Wed, 1 Oct 1997 14:57:44 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <13684FD9F298D011AD8800609769CE770712F1@mailhost.Instantel.com>
Rendered body type: text/plain

Hi there !

 I need a reliable method to measure the NF of a receiver (220 Mhz),
using a HP (8647A) generator,and a
 spectrum analyzer Advantest (R3261A) or a scope (Tektronix TDS 380).

 I know ,there are some devices like HP 8970A (Noise Figure Meter),
but I don't have it.

 Thanks in advance !

  Ted
  613 592 4642

4. Terminal program

From: "Paulus" <paulstra@xs4all.nl>
Date: Wed, 1 Oct 1997 22:28:21 +0100
To: <hfsig@tapr.org>
Message-ID: <199710012029.WAA01943@smtp1.xs4all.nl>
Rendered body type: text/plain

Hello all,

Anthony asked for an terminal program for win95.
I use often Qmodem for win95 , but also Term, which is an Dos program and
runs also fine in an Dosbox in win95.
It works fine for MT63asc.
I believe it is only in the German language available (freeware) , but will
be happy to send it over to you.

Is there more interest for such an program ?, then I can send it to all.

73 de Paul, pa0ocd

paulstra@xs4all.nl

5. Re: [HFSIG:2481] Re: digital test

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 1 Oct 1997 18:28:50 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199710020128.SAA06874@servo.qualcomm.com>
Rendered body type: text/plain

>I think that there are also some unusual phase-change effects at the
>high power levels,

Right. This is also a problem in satellite communications where it's
called "AM-PM conversion" and in TV broadcasting where it's called
"differential phase". In both cases, the phase shift through the tube
is a function of signal level.

During TV transmitter engineering periods you'll sometimes see a test
signal that looks like a shaded greenish-yellow window, varying
continuously in brightness from very dark on one edge to bright at the
other. It's usually part of a "composite test signal" that includes a
wide sharp-edged vertical white bar (a 75% or 100% white reference)
plus a couple of shorter impulses that look like thin vertical white
or multicolored bars.

A single line of this test signal is usually carried as a Vertical
Interval Test Signal (VITS) during normal operation, along with a line
of color bars, closed captioning, etc.

The shaded greenish-yellow window is a linearity test signal that
consists of a constant color component (with the same phase as the
color burst, hence the greenish-yellow color) on top of a linear
luminance ramp. The idea is to detect any intermodulation distortion
from the low frequency luminance signal on either the amplitude or the
phase of the high frequency color subcarrier. This is typically
analyzed by running the signal through a high pass filter (stripping
off the luminance) and looking for variations in amplitude or phase in
the color subcarrier signal that remains.

Because so much work has gone into testing and analyzing the
distortions of NTSC video transmitters, you could do worse than to use
standard video test equipment to test any linear amplifier wide enough
to accomodate an analog video signal. Or you can apply the same
principles to the design of your own test signals. Running scrambled
QAM through an amplifier and looking at the resulting constellation is
also a pretty severe test.

Phil

6. RE: [HFSIG:2487] How to measure Noise Fi

From: Ilkka.Kontola@nmp.nokia.com (Kontola Ilkka NMP)
Date: Thu, 02 Oct 1997 11:39:35 +0200
To: hfsig@tapr.org (hfsig)
Message-ID: <1997Oct02.123400.1711.3069922@sagws01.nmp.nokia.com>
Rendered body type: text/plain

> I need a reliable method to measure the NF of a receiver (220 Mhz),
>using a HP (8647A) generator,and a
> spectrum analyzer Advantest (R3261A) or a scope (Tektronix TDS 380).

> I know ,there are some devices like HP 8970A (Noise Figure Meter),
>but I don't have it.

How about buying only the calibrated noise source such as HP346A and
using your
spectrum as the tunable, selective power meter to measure the Y factor?

(If the LNA gain is not very high you may need a preamplifier for the
SA.)

73
Ilkka OH3NJC (ilkka.kontola@nmp.nokia.com)

7. [HFSIG:2490] RE: How to measure Noise Fi -Reply

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Thu, 02 Oct 1997 13:15:24 +0200
To: hfsig@tapr.org
Message-ID: <s4339e78.014@KIDD.CO.ZA>
Rendered body type: text/plain

Yes but you need a 0.1dB step attenuator not so ?

>>> Kontola Ilkka NMP <Ilkka.Kontola@nmp.nokia.com> 2/October/1997
11:45am >>>

> I need a reliable method to measure the NF of a receiver (220 Mhz),
>using a HP (8647A) generator,and a

> spectrum analyzer Advantest (R3261A) or a scope (Tektronix TDS
380).

> I know ,there are some devices like HP 8970A (Noise Figure Meter),
>but I don't have it.

How about buying only the calibrated noise source such as HP346A
and
using your
spectrum as the tunable, selective power meter to measure the Y
factor?

(If the LNA gain is not very high you may need a preamplifier for the
SA.)

73
Ilkka OH3NJC (ilkka.kontola@nmp.nokia.com)

8. RE: [HFSIG:2491] RE: How to measure Nois

From: Ilkka.Kontola@nmp.nokia.com (Kontola Ilkka NMP)
Date: Thu, 02 Oct 1997 14:27:32 +0200
To: hfsig@tapr.org (hfsig)
Message-ID: <1997Oct02.151800.1711.3072022@sagws01.nmp.nokia.com>
Rendered body type: text/plain

Using a 0.1 dB step attenuator will improve the accuracy since
it allows you to keep the power level constant. In other words
the SA would be used only as some sort of null detector.

For slightly reduced accuracy you can read the Y-factor
directly from the SA. (I do not have any experience concerning
the R3261A ... there is HP 8563E on my desk.)

Ilkka

From: hfsig[SMTP:hfsig@tapr.org]
Sent: 2. lokakuu 1997 6:18
To: hfsig
Subject: [HFSIG:2491] RE: How to measure Noise Fi -Reply

Content-Type: text/plain
X-Mailer: Novell GroupWise 4.1

Yes but you need a 0.1dB step attenuator not so ?

>>> Kontola Ilkka NMP <Ilkka.Kontola@nmp.nokia.com> 2/October/1997
11:45am >>>

> I need a reliable method to measure the NF of a receiver (220 Mhz),
>using a HP (8647A) generator,and a
> spectrum analyzer Advantest (R3261A) or a scope (Tektronix TDS
380).

> I know ,there are some devices like HP 8970A (Noise Figure Meter),
>but I don't have it.

How about buying only the calibrated noise source such as HP346A
and
using your
spectrum as the tunable, selective power meter to measure the Y
factor?

(If the LNA gain is not very high you may need a preamplifier for the
SA.)

73
Ilkka OH3NJC (ilkka.kontola@nmp.nokia.com)

9. Re: [HFSIG:2487] How to measure Noise Figure of a receiver ?

From: Antony Martin <anthony.martin@gecm.com>
Date: Thu, 02 Oct 1997 17:41:29 +0100
To: hfsig@tapr.org
Message-ID: <971002174129.19237@oas01.gmcl.co.uk>
Rendered body type: text/plain

>
> Hi there !
>
> I need a reliable method to measure the NF of a receiver (220 Mhz),
> using a HP (8647A) generator,and a
> spectrum analyzer Advantest (R3261A) or a scope (Tektronix TDS 380).
>
> I know ,there are some devices like HP 8970A (Noise Figure Meter),
> but I don't have it.
>
> Thanks in advance !
>
>   Ted
>   613 592 4642
>
The HP8970 is very difficult to use if there's a frequency conversion
involved. We couldn't use it as it's input BW is very wide and had a
habit of including unwanted signals.

For measuring HF receivers I used a Rhode & Schwarz calibrated noise
source with a variable output level. Measuring the IF output with a
spectrum analyser, you crank up the noise source until the output
increases by 3dB. At this point the noise source output power is equal
to the input noise power of the receiver.

To get repeatable results it was found necessary to do it in a screened
room. This was at HF; your conditions might vary.

You might also be able to use your analyser to measure/compute the noise
level in 1Hz BW; a lot of analysers have buttons for this. If it hasn't
got a button, you would need to know the noise BW of the analyser for the
selected range and all sorts of fiddle factors which makes it a bit hit&miss.
You can then add a small input signal of known level with the sig. gen. and
without changing analyser range read the corresponding output level. From
these readings you can calculate the level of the input noise relative to the
small input signal and hence the noise power. I know someone who swears by
this method but I didn't find it very good.

I suppose it all depends on whether you're looking for a 10 dB NF or a
1 dB NF what accuracy you need.

Anthony.

10. Re: Terminal Program

From: Bob Johnson <wb4jcm@amsat.org>
Date: Fri, 03 Oct 1997 12:29:25 -0400
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19971003122925.006ce400@rio.atlantic.net>
Rendered body type: text/plain

> HFSIG Digest 419

> 2) Terminal program

>Subject: Terminal program

>Anthony asked for an terminal program for win95.[...]

I've lost the original posting, but as I recall, his description
sounded like his problem was hardware, not software. I've never had
trouble using HyperTerminal or ProComm at speeds up to 110KB.

Make sure you have a high-speed (68550 or equivalent IC) serial
port in your computer. The older (original) serial ports
weren't really designed to operate over 9600 bps, although
most could, depending on the speed of the CPU. Using a newer
serial card should make even a '286 work reliably at high speed.

Even my expensive local computer store sells fast serial cards
for about $25, so the upgrade (if that's the problem) should
be relatively painless.

-- Bob
   wb4jcm@amsat.org
   bobj@atlantic.net

-- Bob Johnson

Above comments are personal and are not intended to reflect
the official position of any other person or corporation.

11. RE: [HFSIG:2485] Re: TMS320C50 version for PSK31

From: "Louis Cobet" <Louis_Cobet@classic.msn.com>
Date: Sun, 5 Oct 97 01:58:32 UT
To: hfsig@tapr.org
Message-ID: <UPMAIL14.19971005015900019l@classic.msn.com>
Rendered body type: text/plain

Anthony Martin wrote:

>Can anyone recommend a *good* 'terminal emulator' program for >windows 95?

For personal use, I like Qmodem for windows 95. You can buy it for $25 or $50
from Mustang software. It has a fairly good scripting language.

Its downloadable from the Mustang web site: http://www.mustang software.

Louis Cobet, K6MDH

Louis_Cobet@msn.com

12. Ringing in CW filters

From: zs6awk@global.co.za (Danie Brynard)
Date: Fri, 03 Oct 1997 09:54:00 +0200
To: hfsig@tapr.org
Message-ID: <199710050539.HAA08909@mail.global.co.za>
Rendered body type: text/plain

Please some comments:

Is ringing a fact of life for vy narrow CW filters or can one design a type
that will have none ?

I have always believed that narrow high Q filters must ring, true ?

(My question is sparked by a comment from a friend who said that his narrow
CW filters has no ringing)

Danie

++
Danie Brynard Email: zs6awk@global.co.za
PO Box 15133,Lyttelton,0140,South Africa
GRID LOC: KG44CE, Tel/Fax: (2712) 6640330
++

13. Re: [HFSIG:2496] Ringing in CW filters

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Mon, 6 Oct 1997 13:38:41 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199710061038.NAA00216@boojum.hut.fi>
Rendered body type: text/plain

This is so general, that even I can comment... ;-)

> Is ringing a fact of life for vy narrow CW filters or can one design a type
> that will have none ?
>
> I have always believed that narrow high Q filters must ring, true ?
>
> (My question is sparked by a comment from a friend who said that his narrow
> CW filters has no ringing)

If you write the equation of motion of any oscillator (mechanical spring-
mass or electrical LC-resonator) you notice that there is a damping term
(describing the Q-value) and an external force term. If we build an
oscillator the external force must compensate the losses in the resonator.
There is always a noise source present, however. This will give a random
force to the oscillator and will excite the degrees of freedom allowed
by the system. For a high Q-resonator (which maybe a part of a narrow band
filter) this means that the wideband noise becomes narrowband hissing.
A useful measure is the rise time of the resonator t = 2 Pi f/Q.
It gives you idea on how fast the signal amplitude in the resonator
can change. For times less than t your resonator is just oscillating
and you can hear a continuous slowly decaying tone.

There are other narrow band techniques available where you do not need
to give the system such degree of freedom for oscillation. A phase
coherent detection for example.

--
Jaakko Koivuniemi/OH7BF

14. Re: [HFSIG:2496] Ringing in CW filters

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Mon, 6 Oct 1997 14:25:42 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199710061125.OAA00760@boojum.hut.fi>
Rendered body type: text/plain

I don't remember how did I write the rise time of a resonator. It shoud be
t= 2 Pi Q /f. The higher Q - the longer ringing time. Sorry for the mistake.

--
Jaakko Koivuniemi/OH7BF

15. [HFSIG:2498] Re: Ringing in CW filters -Reply

From: Danie Brynard <BRYD@KIDD.CO.ZA>
Date: Mon, 06 Oct 1997 14:04:22 +0200
To: hfsig@tapr.org
Message-ID: <s438effa.014@KIDD.CO.ZA>
Rendered body type: text/plain

resonator t = 2 Pi f/Q.   Thats what you wrote. Thanks for the nice
answer. Danie zs6awk

>>> Jaakko Koivuniemi <jaakko@boojum.hut.fi> 6/October/1997 01:26pm
>>>

I don't remember how did I write the rise time of a resonator. It shoud be
t= 2 Pi Q /f. The higher Q - the longer ringing time. Sorry for the mistake.

--
Jaakko Koivuniemi/OH7BF

16. Re: [HFSIG:2496] Ringing in CW filters

From: Antony Martin <anthony.martin@gecm.com>
Date: Mon, 06 Oct 1997 18:25:50 +0100
To: hfsig@tapr.org
Message-ID: <971006182550.5910@oas01.gmcl.co.uk>
Rendered body type: text/plain

> Please some comments:
>
> Is ringing a fact of life for vy narrow CW filters or can one design a type
> that will have none ?
>
> I have always believed that narrow high Q filters must ring, true ?
>
> (My question is sparked by a comment from a friend who said that his narrow
> CW filters has no ringing)
>
> Danie
>

All high-Q (narrow) filters exhibit ringing to some degree.

The filter, because it is narrow, removes frequency components
in the signal. 'Ringing' is the time-domain representation of this.

Narrower filters or steeper rolloff filters have higher Q and ring
more. This is because they truncate more harmonics and attenuate
them more. If you want a filter that doesn't ring, you have to make
compromises, by making it wider or slower rolloff (like butterworth).

Overshoot etc. which occur on a waveform near discontinuities which
result from the truncation of the fourier series are known as the
'Gibbs phenomenon'.

Filters such as L-C and digital IIR are resonant and can be said to
have Q. FIR filters have no feedback and thus cannot have 'Q' and
can't be said to ring in the same way. But they are no less filters
and will still create unwanted time-domain effects, but as they're
not exponentially decaying to infinity can't truly be described as

'ringing'. They have a finite decay time, hence 'Finite Impulse
Response'.

The frequency-domain response (amplitude & phase) of the filter
translates into a time-domain response (ie. distorted edges) which
you may not like. It isn't dependent on how that response is
implemented in physical components.

Precisely what constitutes 'Ringing' seems to be a bit unclear in
digital filters... 'Ringing' implies resonance which implies a
filter with poles. FIR filters are all-zero. A salesman might then say
'it doesn't ring'. But time-domain pulse distortion will still be
there, it might look a bit like ringing so is it ringing???

Anthony.

17. Re: [HFSIG:2495] Re: TMS320C50 version for PSK31

From: Ulf.Kumm@t-online.de (Ulf Kumm)
Date: Tue, 07 Oct 1997 15:22:43 +0200
To: hfsig@tapr.org
Message-ID: <343A37A3.31B1@t-online.de>
Rendered body type: text/plain

> Anthony Martin wrote:
>
> >Can anyone recommend a *good* 'terminal emulator' program for >windows 95?
>
Try uploading

http://members.aol.com/mitlaender/wingt.htm

Gerd Mitländer describes the 'WinGT' program which is excellent for
hostmode ('WA8DED') and PC-Windows.

73! Ulf
--
Ulf Kumm, DK9SJ, Mail: ulf@symek.com
D-70597 Stuttgart,   Johannes-Kraemer-Strasse 34
Phone: +49 711 7678-925, Fax: -924, http://symek.com

18. AMRAD meeting Thur of DCC

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Tue, 7 Oct 1997 09:59:17 -0500
To: "TAPR-BB list   mailing"<tapr-bb@tapr.org>, " tacgps "<tacgps@tapr.org>,
        " Spread Spectrum "<ss@tapr.org>,
        "NETSIG list   mailing"<netsig@tapr.org>,
        "HF SIG list   mailing"<hfsig@tapr.org>
Message-ID: <v03110708b05ffe1db177@[208.134.134.40]>
Rendered body type: text/plain

Participants in the 1997 Digital Communications Conference are invited to
attend the regular monthly meeting of the Amateur Radio Research and
Development club (AMRAD), Thursday evening at 7:30 PM in the lower meeting
room of the Dolley Madison Branch Libary, McLean, Virginia. The library is
located near the intersection of VA Rte 123 (Dolley Madison Blvd) and Old
Dominion Drive.

This month's meeting is devoted to the "dc" band of anywhere from 73 kHz to
190 kHz. ARRL has stated an amateur requirement for access to a
low-frequency band and NTIA has accepted the justification. There are still
many steps needed to acquire the band but interest is building among
experimenters. American LF advocates are watching activities in Europe
where CEPT is planning a "harmonised" band around 143 kHz.

Bob Laney will be one of the (informal) speakers for this Thursdays
meeting. He has a wonderful LF set-up and is even a fan of amateur
seismology. He'll be bringing photos and share his experiences in the LF
arena.

19. EZ-KIT SHARC progress

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 07 Oct 1997 08:57:14 -0800
To: DSP@tapr.org, HFSIG@tapr.org
Message-ID: <199710071557.IAA03019@peak.org>
Rendered body type: text/plain

Hi,

Thought I'd share some progress made on the EZ-KIT SHARC. If you don't know

this evaluation board; it contains a 40MHz ADSP-21061, 32-bit floating point DSP, and the kit comes with a C compiler and other development tools (restricted for the kit, but otherwise quite functional). I payed $189 a few months ago when I got it.

I now have gotten my HF channel simulator working on this platform. Just to see whether it could be done, I programmed it all in C, stood back to watched what happened. Well, surprize - it works just fine at the 8kSPS rate that I choose! A minor amount of code tweaking was necessary to take non-critical stuff out of critical paths, and in some cases I had to unroll a few function calls to save on calling overheads. But I did not even use cicular buffers - instead are using regular C pointer arithmetic. This means that if one would carefully write in mixed C and assembly to streamline some of the classical DSP operations like FIR filters, you could actually do a lot better. Seems like this GNU port for the SHARC does a fairly decent job.

The simulator does everything in 40-bit floating point arithmetic; the HF channel is modeled after the Watterson ionospheric model where the channel is implemented as an analytic signal (complex tapped delay line). One direct and one delayed ray is taken from the delay line, Rayleigh fading is introduced and finally mixed with a noise model. The Rayleigh fading model requires a very good random number generator (six calls to this one alone), sine/cosine function calls (four calls to these two), IIR lowpass filtering (four times this one is called) and two complex mixers. The resultant dynamic range is wonderful - I can now observe affects that previously was missed with fixed point DSP.

Althought I still am a firm believer that it makes sense to write DSP-specific code in assembly, this example illustrates that one can be quite productive using C and a chip like this. Although I'm still learning to drive this DSP, looks like it may become a regular choice for my future DSP projects.

73

--Johan, KC7WW

20. Re: [HFSIG:2502] AMRAD meeting Thur of DCC

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 07 Oct 1997 17:11:40 +0100
To: hfsig@tapr.org
Message-ID: <343A5F3C.11DF6F87@dircon.co.uk>
Rendered body type: text/plain

Greg Jones, WD5IVD wrote:

> experimenters. American LF advocates are watching activities in Europe
> where CEPT is planning a "harmonised" band around 143 kHz.
>

The U.K will be getting a band at 136 khz starting next January, the 73 Khz band is being phased out two years after the new harmonised band comes online.

- Charles G4GUO

21. Re: [HFSIG:2504] Re: AMRAD meeting Thur of DCC

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Tue, 7 Oct 1997 14:09:12 -0500
To: hfsig@tapr.org
Message-ID: <v03110715b06039459aea@[208.134.134.40]>
Rendered body type: text/plain

Thanks Charles.

I'll bring this up at the ARRL Future Systems Committee Meeting this weekend.

Cheers - Greg

>Greg Jones, WD5IVD wrote:
>
>> experimenters. American LF advocates are watching activities in Europe
>> where CEPT is planning a "harmonised" band around 143 kHz.
>>
>
>The U.K will be getting a band at 136 khz starting next January, the
>73 Khz band is being phased out two years after the new harmonised
>band comes online.

>
>
>- Charles G4GUO

Greg Jones, WD5IVD
Austin, Texas
wd5ivd@tapr.org
http://www.tapr.org/~wd5ivd

22. Re: [HFSIG:2504] Re: AMRAD meeting Thur of DCC

From: Tomi Manninen <tomannin@trshp.ntc.nokia.com>
Date: Wed, 8 Oct 1997 12:21:04 +0300 (EETDST)
To: hfsig@tapr.org
Message-ID: <Pine.HPP.3.95.971008121736.13947A-100000@jas.trs.ntc.nokia.com>
Rendered body type: text/plain

On Tue, 7 Oct 1997, Charles Brain wrote:

> Greg Jones, WD5IVD wrote:
>
> > experimenters. American LF advocates are watching activities in Europe
> > where CEPT is planning a "harmonised" band around 143 kHz.
>
> The U.K will be getting a band at 136 khz starting next January, the
> 73 Khz band is being phased out two years after the new harmonised
> band comes online.

We Finns have had access to 135.7 - 137.8 kHz since April 1 this year. See
http://www.thk.fi/saados/englanti/thk0397.htm.

--
---Tomi Manninen / Tomi.Manninen@hut.fi / OH2BNS @ OH2RBI.FIN.EU---

23. RE: SHARC DSP for HF

From: "Furman, William" <WFurman@harris.com>
Date: Wed, 8 Oct 1997 07:09:00 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <51650CE89911D111B4540000F840E3981699FE@rfcmx2.rfc.comm.harris.com>
Rendered body type: text/plain

Hello All -

Just wanted to mention that the SHARC C compiler includes an FIR

function that allows you to implement a FIR filter in 1 cycle / tap (excluding setup).
This is accomplished by placing the tap coefficients in program memory.

Regards,

Bill Furman

24. Re: [HFSIG:2500] Re: Ringing in CW filters

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Wed, 08 Oct 1997 11:51:12 -0400
To: hfsig@tapr.org
Message-ID: <199710081551.LAA12408@aus-f.mp.campus.mci.net>
Rendered body type: text/plain

Anthony,

Nice reply to the "ringing" question. It is something that I've pondered over the years. I remember debating whether a narrow band band 455 Khz IF filter rang more than the same bandwidth implemented with a 5 Mhz xtal IF filter, with an old-timer ham-radioman in the Navy, circa 1959 (my conclusion: essentially no difference; his position: low freq filters rang more). You raise an interesting issue--precisely what is "ringing."

I've always thought of ringing in terms of the effect of lightning crashes when listening to CW on the low bands. I'm not sure what "ringing" might be in terms of slow BPSK (on HF).

Classically I guess one would say that if the filter is given an impulse for an input, the ringing is the response, i.e. the impulse response of the filter. For a simple high-Q tuned circuit an impulse produces a negative

exponential decay of sine wave at the resonant frequency. For a FIR filter
an impluse simply "reads out" the coefficients of the filter as the impulse
value shifts with each sample, i.e. the coefficients of an FIR filter is the
impulse response. This is not an equal comparison, however. Better would
be a recursive filter (IIR) versus FIR filter with equal bandwidths, at say,
3 db and 30db points. I'm not sure which produces more "ringing."

If the criteria for "ringing" could be stated then I supposed one could
devise a procedure for optimizing bandwidth & rolloff with "ringing." BTW,
the Butterworth criteria is "maximally flat" (in the passband) for which the
slower rolloff is an artifact. The derivation applies to recursive, i.e.
IIR filters. R.W. Hamming (_Digital Filters_) shows an equivalent
derivation for FIR filters.

Intuitively it appears that the energy in the impluse can only be "mushed"
into either short & high, or long & low responses--the total energy is not
affected by the filter (other than the scaling of all signals). What is
more objectionable would depend on the demodulation scheme, i.e. "ear" for
cw, or matched filter for BPSK. This line of reasoning would suggest that
implementing a "noise-blanker" (in DSP) might help, at least under some
comditions, (e.g. "detect" a broad-band impluse and input the last sample
before the impulse to the filter). This issue is related to earlier
thread, "Static Crashes and FEC." I wonder if anyone has seen any research
or articles on "dsp noise-blankers." Possibly it is comes under a different
title, such as "threshold decisions" or some such jargon.

73's
Don, W4DH

25. Re: [HFSIG:2508] Re: Ringing in CW filters

From: Keith McElveen <jkm@mclean.sparta.com>
Date: Wed, 08 Oct 1997 13:35:25 -0400
To: hfsig@tapr.org
Message-ID: <343BC45D.6A6@mclean.sparta.com>
Rendered body type: text/plain

Seems like some type of smoothing filter is called for. Perhaps a low
pass or a median filter (take the median value of 'x' samples) might do
the trick without costing too much in the way of processor resources.

>This line of reasoning would suggest that
> implementing a "noise-blanker" (in DSP) might help, at least under some
> comditions, (e.g. "detect" a broad-band impluse and input the last sample
> *before* the impulse to the filter). This issue is related to earlier
> thread, "Static Crashes and FEC." I wonder if anyone has seen any research
> or articles on "dsp noise-blankers." Possibly it is comes under a different
> title, such as "threshold decisions" or some such jargon.

26. Re: [HFSIG:2508] Re: Ringing in CW filters

From: Antony Martin <anthony.martin@gecm.com>
Date: Wed, 08 Oct 1997 22:13:12 +0100
To: hfsig@tapr.org
Message-ID: <971008221312.3023@oas01.gmcl.co.uk>
Rendered body type: text/plain

>
> If the criteria for "ringing" could be stated then I supposed one could
> devise a procedure for optimizing bandwidth & rolloff with "ringing." BTW,
> the Butterworth criteria is "maximally flat" (in the passband) for which the
> slower rolloff is an artifact. The derivation applies to recursive, i.e.
> IIR filters. R.W. Hamming (_Digital Filters_) shows an equivalent
> derivation for FIR filters.

I assert that two filters with identical frequency responses must have
identical impulse responses. The fact that different filter
implementations have different phase responses while they have similar
amplitude responses affects the impulse response to some degree, but
I think the difference is not enough to be important. Symmetrical FIR
filters tend to ring on BOTH sides of the impulse, although probably
half as much on each! Does that matter?
>
> Intuitively it appears that the energy in the impluse can only be "mushed"
> into either short & high, or long & low responses--the total energy is not
> affected by the filter (other than the scaling of all signals). What is

You are very right. For a filter to pass the wanted signal with unity

gain, the impulse response MUST also sum to unit energy, whatever it's shape.

There's a subtle difference though between the analogue impulse and the digital sampled-system equivalent. An analogue impulse has infinite amplitude and zero width, and has unit energy. A digital impulse has 1 sample period width and unit amplitude. This is in fact the perfect representation of the analogue impulse band-limited by a brick-wall anti-aliasing filter to fit into the nyquist rate!

If you think about this a little, it may dawn on you... if it's been band-limited by a filter, then why is there no ringing?!!! Answer: the zero crossings of the ringing are at the sampling instants so the other sample values are zero. Unfortunately real ADCs don't have perfect anti-aliasing filters, so some ringing may be evident.

> more objectionable would depend on the demodulation scheme, i.e. "ear" for
> cw, or matched filter for BPSK. This line of reasoning would suggest that
> implementing a "noise-blanker" (in DSP) might help, at least under some
> comditions, (e.g. "detect" a broad-band impluse and input the last sample
> *before* the impulse to the filter). This issue is related to earlier
> thread, "Static Crashes and FEC." I wonder if anyone has seen any research
> or articles on "dsp noise-blankers." Possibly it is comes under a different
> title, such as "threshold decisions" or some such jargon.
>

The problem with impulses is that they will already have been BW limited and clipped by the receiver IF before the DSP ever sees them, and thus may already be 'rung' and will carry the impulse response of your receiver. It's no longer just a spike any more.

By the time the spike has been averaged with all the other samples in the symbol-period, it ammounts to a small net change in the general noise floor. Unless you're using short symbols I doubt it's worth doing anything.

Anthony.

> 73's
> Don, W4DH
>
ps.

Does anyone know how non-symmetrical complex FIR filters and lattice filters can be designed? My textbooks don't help. I can feel a very nasty filter problem coming on real soon.

27. Re: [HFSIG:2510] Re: Ringing in CW filters

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 8 Oct 1997 16:58:53 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199710082358.QAA06384@servo.qualcomm.com>
Rendered body type: text/plain

I've been meaning to comment on this thread, but I've been busy preparing for my SS tutorial at the upcoming TAPR/ARRL CNC.

>I assert that two filters with identical frequency responses must have
>identical impulse responses. The fact that different filter

Not true. For the impulse responses to be the same, the frequency responses *and* the group delay (phase) responses MUST be the same.

A filter's impulse response is symmetric if and only if the group delay is constant for all frequencies (i.e., the filter is "linear phase"). Since it's easy to generate symmetric impulse responses with FIR filters, and since constant group delay is usually considered a desirable property in a filter, most FIR filters are designed to exhibit symmetric impulse responses (i.e., constant group delay or linear phase).

One major exception is an FIR filter designed specifically to correct for a phase nonlinearity someplace else, as on a communications channel, to make the overall response linear phase. These are called "equalizing filters". Another is the "Hilbert transformer" used in phasing SSB generation. It has an ANTI-symmetric impulse response, which is as non-linear-phase as you can get. :-)

IIR filters, on the other hand, are more like digital simulations of
analog filters. They may have "better" characteristics (sharper
skirts, etc) than a FIR filter of the same computational complexity,
but this comes at the expense of uncontrolled group delay (just as in
analog filters). For this reason, IIR filters are generally limited to
applications where CPU cycles are limited and group delay is
relatively unimportant, such as speech synthesis and vocoding (the
human ear being relatively insensitive to group delay). FIR filters
are strongly preferred in modems where group delay is usually very
important.

"Ringing" is an inherent feature of *any* narrow, sharp-skirted
filter. It's not due to imperfect components, it's basic to the
mathematics. Because frequency and time are complementary, the
narrower the frequency response, the wider the time response and vice
versa. It's as fundamental to filter theory as the Heisenberg
Uncertainty Principle is to physics. In fact, they're very closely
related.

Note that any realizable filter cannot predict the future of its
input. So the entire response (including the "pre-ringing" before the
peak response) must occur after the input impulse that triggers
it. This can imply substantial propagation delay if the impulse
response is long, which it necessarily is if the filter skirts are
sharp. Most FIR filters have a group delay equal to one half the delay
line length. The longer the filter, the sharper the response that is
possible but the longer the group delay.

Note also that it is neither necessary nor desirable to eliminate
ringing in a filter to be used for digital communications. It is
sufficient merely to design it such that the ringing "cycles" from
previous and future bits all go through zero when the current bit is
sampled. You don't care what happens between bits.

It is both necessary and sufficient to accomplish this with a channel
response that is a) linear phase and b) has an amplitude response that
is anti-symmetric about the frequency equal to one half of the data
rate (the Nyquist rate).

This is the theory behind the "square root of raised cosine" filters
that you can find discussed in N5EG's book, and has also been
discussed on this list in the past. The raised-cosine channel response
function satisfies the anti-symmetric requirement. And taking the
square root of this function at each end allows the transmit and
receive filters to be matched, giving the best performance possible on
a noisy channel.

Phil

28. Re: [HFSIG:2511] Re: Ringing in CW filters

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Wed, 08 Oct 1997 23:50:17 -0400
To: hfsig@tapr.org
Message-ID: <199710090350.XAA11766@aus-a.mp.campus.mci.net>
Rendered body type: text/plain

>IIR filters, on the other hand, are more like digital simulations of
>analog filters. They may have "better" characteristics (sharper
>skirts, etc) than a FIR filter of the same computational complexity,
>but this comes at the expense of uncontrolled group delay (just as in
>analog filters). For this reason, IIR filters are generally limited to
>applications where CPU cycles are limited and group delay is
>relatively unimportant, such as speech synthesis and vocoding (the
>human ear being relatively insensitive to group delay). FIR filters
>are strongly preferred in modems where group delay is usually very
>important.

An item I ran across (and forgotten) in the RW Hamming book was that if one
runs the data through an IIR filter, then runs through again, but in reverse
order, the phase non-linearity is removed (and the amplitude response is the
filter response squared). For most modem applications that cannot be done,
however it opens up some interesting possiblities when there is there is an
ARQ scheme (where the data samples for the whole block could be stored).

>This is the theory behind the "square root of raised cosine" filters
>that you can find discussed in N5EG's book, and has also been
>discussed on this list in the past. The raised-cosine channel response
>function satisfies the anti-symmetric requirement. And taking the
>square root of this function at each end allows the transmit and

>receive filters to be matched, giving the best performance possible on
>a noisy channel.

The analyses that I remember are all based on gaussian noise. The impulse noise seems to be a different animal, though I suppose that over enough time (and therefore samples) the statistics of impulse noise approximates a gaussian curve (i.e the central limit theorem); it's just that the time tends to be very long in terms of symbols.

>Phil
>

In rough terms it would seem that the matched filter would result in "spreading" an impulse to about that of the symbol period. If the signal-to-gaussian noise is "good" but the impulse noise is orders-of-magnitude greater than the signal, then (and without a noise-blanker) I would think that a filter that is wider than the matched filter would be better, since it would narrow the filter response to the impulse.

I suppose one could view the ADC and saturated arithemtic as a form of noise blanker as well. If the impulse noise is orders of magnitude greater than the signal, then a spike results in first a max ADC output (which is not orders-of-magnitude greater than the signal) and possibly the computations going into overflow.

Anthony makes a good point about the receiver's early stages also "ringing." This would suggest getting a noise-blanker early in the receive chain would be best. However, I suspect the problem is that something like a nearby, very strong CW signal (which is outside the final filter passband) would result in the blanking the desired signal. I've observed this on my FT 757GX. Turning off the noise-blanker cleans up the desired signal.

Thinking it through one can see the benefit of coding. (Though computing coding on one's head whilst sending CW with a keyer at 20 wpm would indeed be quite a skill!...I think I'll stick with BPSK & computers.)

73's
Don, W4DH

29. Re: [HFSIG:2511] Re: Ringing in CW filters

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Thu, 9 Oct 1997 11:22:14 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199710090822.LAA02710@boojum.hut.fi>
Rendered body type: text/plain

Thanks for educating us again Phil!

> Note that any realizable filter cannot predict the future of its
> input. So the entire response (including the "pre-ringing" before the

This reminds me that the thread has been mostly on linear filters.
If a neural network can be considered as a "filter" it is usually
actually trained to predict its input. The human ear is one example
of such perception with a neural network. Certainly it will also get out
of focus if there is a sudden impulse in the CW reception. But it does not
"ring" in the classical sense. The mixers in receivers and parametric
amplifiers could also be understood as a kind of "non-linear filters".
With a parametric amplifier you can actually increase the energy of
the detected signal for a certain phase.

Jaakko/OH7BF
Disclaimer: all the information above is likely to be inaccurate.

30. Re: [HFSIG:2512] Re: Ringing in CW filters

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 9 Oct 1997 01:35:08 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <199710090835.BAA09425@servo.qualcomm.com>
Rendered body type: text/plain

>An item I ran across (and forgotten) in the RW Hamming book was that if one
>runs the data through an IIR filter, then runs through again, but in reverse
>order, the phase non-linearity is removed (and the amplitude response is the
>filter response squared). For most modem applications that cannot be done,
>however it opens up some interesting possiblities when there is there is an

>ARQ scheme (where the data samples for the whole block could be stored).

That is indeed interesting, and practical for any packet system where
the decoding can occur after the entire packet has been received. Not sure
it really helps, though, as FIR filtering is entirely practical thanks to
modern CPU speeds.

>The analyses that I remember are all based on gaussian noise. The impulse
>noise seems to be a different animal, though I suppose that over enough time
>(and therefore samples) the statistics of impulse noise approximates a
>gaussian curve (i.e the central limit theorem); it's just that the time
>tends to be very long in terms of symbols.

Right. Gaussian noise is so common in nature precisely because it's what
you get when you sum a whole bunch of independent random processes. It
hardly even matters what the individual processes look like -- their sum
still looks Gaussian.

>This would suggest getting a noise-blanker early in the receive chain would
>be best. However, I suspect the problem is that something like a nearby,
>very strong CW signal (which is outside the final filter passband) would
>result in the blanking the desired signal. I've observed this on my FT
>757GX. Turning off the noise-blanker cleans up the desired signal.

Right. Still, noise blanking is useful if you can do it intelligently.
Respond only to sharp, wideband pulses, not adjacent channel signals.

Blanking before the matched filter helps to select against the impulse
noise, since its energy is presumably concentrated over a relatively
short time compared to your symbol interval. If you do it well enough,
you may even be able to recover the symbol that got hit by the impulse.

Another benefit of blanking is to keep a slow-decay AGC from being
driven down beyond the end of the pulse.

It may be that with coding to fill in data hit by an impulse, and with
a fast-acting AGC (preferably with the AGC setting being carried
through as a scale factor or exponent in each A/D sample so soft
decision decoding can work), blanking isn't as essential as it used to
be. But it's still worth trying if impulse noise is common.

Phil

31. Re: [HFSIG:2512] Re: Ringing in CW filters

From: Antony Martin <anthony.martin@gecm.com>
Date: Thu, 09 Oct 1997 15:30:14 +0100
To: hfsig@tapr.org
Message-ID: <971009153014.10856@oas01.gmcl.co.uk>
Rendered body type: text/plain

>
> >IIR filters, on the other hand, are more like digital simulations of
> >analog filters. They may have "better" characteristics (sharper
> >skirts, etc) than a FIR filter of the same computational complexity,
> >but this comes at the expense of uncontrolled group delay (just as in
> >analog filters). For this reason, IIR filters are generally limited to
> >applications where CPU cycles are limited and group delay is
> >relatively unimportant, such as speech synthesis and vocoding (the
> >human ear being relatively insensitive to group delay). FIR filters
> >are strongly preferred in modems where group delay is usually very
> >important.

I have been reading about the 'minimum phase' FIR filters. In many
comms systems the long delay of the linear-phase FIR filter is
unacceptable. The minimum phase offers very little delay. The phase
response isn't flat but it's nearly flat for most of the passband.
Minimum and Maximum phase filters may be half the length of
linear phase FIR if the passband is wide.

FIR are good for decimation as you don't calculate anything for output
samples you don't need. If IIR is used for decimation you still have to
do all the calculations, even though you don't want the output sample.
FIR also allows use to be made of the wider accumulators of the DSP to
avoid rounding errors at intermediate stages of the calculation.

IIRs are usually designed as cascaded sections to allow analysis of
stability, limit-cycles and saturation oscillation problems. As a
result, they don't code well on DSPs designed for efficient repetition.
Multiple precision arithmetic may also be needed to implement a narrow IIR.

I've been deafened several times by the excruciating noise that comes
out of the handset/headphones when an IIR goes into saturation
oscillation; very difficult to control; just forgetting to initialise
the internal variables can do it.

>
> >This is the theory behind the "square root of raised cosine" filters
> >that you can find discussed in N5EG's book, and has also been
> >discussed on this list in the past. The raised-cosine channel response
> >function satisfies the anti-symmetric requirement. And taking the
> >square root of this function at each end allows the transmit and
> >receive filters to be matched, giving the best performance possible on
> >a noisy channel.
>

The derivation of the nyquist response filters (of which the raised-cosine
family is *one* example) requires the following:

Linearly-additive White Gaussian noise
Linear time-invariant channel
Zero ISI is desired
Minimum BER is desired
SNR is a constant
A fixed decision threshold

Therein lies the prroblem.

HF noise isn't gaussian. Phase noise isn't white or additive.
Channels aren't time-invariant; there's fading and doppler.
Channels aren't linear; PAs saturate, and frequency-errors are not linear.
It may be desirable to trade ISI, eg. for lower delay.
You may want to optimise something other than BER; eg recovered clock.
SNR may depend on transmitted waveform(& hence filter shape).
There may be no fixed threshold (soft decision).

Interferers have been entirely neglected.
Transmitter power efficiency has been neglected.
Bandwidth efficiency has also been neglected.

It looks like real life has departed a long way from the assumptions.
If the assumptions were changed would it make any difference???

As there are infinite variations of possible filters which meet the
nyquist criteria, is it not likely that additional assumptions might
lead us to choose many different designs of nyquist filter?

A FIR filter that meets the nyquist criteria simply has every
Nth coefficient zero except the middle one, where N is the
number of samples per symbol. You have all the other coefficients
free for you to choose according to your wishes, and thus
can have almost any amplitude and phase response that you want.

On the basis that in-band energy distribution should be flat
and that out-of-band energy should be minimised IMHO the
filter should be as brick-wall as possible, but your
opinions may be just as good.

The raised-cosine is an anachronism belonging to the days of analogue
circuits. We now use DSPs; we can have any filter we like. Why
pay lip-service at the shrine of the raised-cosine? It needs to be
well and truly buried, dead, and forgotten and of only passing interest
to historians, and no interest to engineers at all.

Anthony

32. Re: [HFSIG:2513] Re: Ringing in CW filters

From: Phil Karn <karn@qualcomm.com>
Date: Fri, 10 Oct 1997 11:46:33 -0700 (PDT)
To: hfsig@tapr.org
Message-ID: <m0xJk53-000HQ7C@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>If a neural network can be considered as a "filter" it is usually
>actually trained to predict its input. The human ear is one example

This is a different kind of "prediction" than I had in mind. When I
said that a filter cannot predict the future, I meant this in the
general sense that causality must be obeyed. By definition, something

that is unpredictable cannot be predicted! (Who was it who said that
predictions are always hard, especially about the future?)

But in many real systems, the future (especially the immediate future)
is a strong function of the past. Then prediction does become
possible. Neural networks can do this, as can, for example, speech
coders. Linear Predictive Coding works by looking at the past
properties of the speech, projecting ("predicting") them into the
future, and transmitting the difference between the predicted and
actual values. Obviously if the predictions were absolutely reliable,
there would be no need to transmit anything!

Phil

33. Re: [HFSIG:2516] Re: Ringing in CW filters

From: Jaakko Koivuniemi <jaakko@boojum.hut.fi>
Date: Mon, 13 Oct 1997 11:35:51 +0300 (EET DST)
To: hfsig@tapr.org
Message-ID: <199710130835.LAA00229@boojum.hut.fi>
Rendered body type: text/plain

> But in many real systems, the future (especially the immediate future)
> is a strong function of the past. Then prediction does become
> possible. Neural networks can do this, as can, for example, speech
> coders. Linear Predictive Coding works by looking at the past

Yes. Instead of locking to a signal you lock to the information, a kind
of "information phase-locked loop". Sorry for this very general discussion,
when will I have time to do something real?

--
Jaakko Koivuniemi/OH7BF
Disclaimer: all the information above is likely to be inaccurate.

34. Infra-red MD problem

From: Herb Salls WB1DSW <whftherb@nh.ultranet.com>
Date: Mon, 13 Oct 1997 15:36:21 -0700
To: hfsig@tapr.org
Message-ID: <3442A265.5478@nh.ultranet.com>
Rendered body type: text/plain

Hello all.

This doesn't really pertain to DSP or SS or much of the other esoteric
topics I see here. Let me just pose this little situation I have with
the hope that someone out there may have run across this before.

I operate a full-time HF forwarding system (CLOVER-II) just north of the
Boston area. At night, I switch the focus of the operation to 40
meters.
I have just recently installed infra-red motion detectors on the outside
lighting. Whenever the 40 meter system fires up, I have one MD that
constantly triggers on. It stays on for the full four minutes (the
threshold I have set) then extinguishes only to re-trigger when and if
the 40 meter transmitter is keyed. The 40 meter system consists of a
Kenwood transceiver, and an all-band dipole fed through a tuner.

While I'm mildly worried about the electric bill, I'm more concerned
about how this looks to the folks who live around me. I've tried moving
the antenna away from it, though I can't say that it's any great
distance (less than a foot). I've tried changing the the MD and even
the brand of MD. I've tried another HF transciever. I've tried calling
the MD manufacturer (Stanley Works) - needless to say they weren't too
sympathetic. I've run out of simple fixes that I can think of.

Do we have any other "great thinkers" out there who might drop me a hint
as to what to try next?

If this is not the correct forum for this kind of question, I'd like a
referral if you know.

Thanks for any consideration.

Herb, WB1DSW.NH.USA.NOAM
whftherb@nh.ultranet.com

35. Re: [HFSIG:2518] Infra-red MD problem

From: Keith McElveen <jkm@mclean.sparta.com>
Date: Mon, 13 Oct 1997 16:40:53 -0400
To: hfsig@tapr.org
Message-ID: <34428755.4BEB@mclean.sparta.com>
Rendered body type: text/plain

Well, I'm hardly a "great thinker" but here go some weak musings.....

Assuming that the MD circuitry may be susceptible to radiated energy
from your antenna (RFI), try shielding and/or grounding the MD box.
Next thought would be susceptibility to conducted radiation, ie, through
the power line, which would require filtering your power and ground
lines going to the MD.

regards,

Keith McElveen

> Hello all.
>
> This doesn't really pertain to DSP or SS or much of the other esoteric
> topics I see here. Let me just pose this little situation I have with
> the hope that someone out there may have run across this before.
>
> I operate a full-time HF forwarding system (CLOVER-II) just north of the
> Boston area. At night, I switch the focus of the operation to 40
> meters.
> I have just recently installed infra-red motion detectors on the outside
> lighting. Whenever the 40 meter system fires up, I have one MD that
> constantly triggers on. It stays on for the full four minutes (the
> threshold I have set) then extinguishes only to re-trigger when and if
> the 40 meter transmitter is keyed. The 40 meter system consists of a
> Kenwood transceiver, and an all-band dipole fed through a tuner.
>
> While I'm mildly worried about the electric bill, I'm more concerned
> about how this looks to the folks who live around me. I've tried moving
> the antenna away from it, though I can't say that it's any great
> distance (less than a foot). I've tried changing the the MD and even
> the brand of MD. I've tried another HF transciever. I've tried calling
> the MD manufacturer (Stanley Works) - needless to say they weren't too
> sympathetic. I've run out of simple fixes that I can think of.
>
> Do we have any other "great thinkers" out there who might drop me a hint
> as to what to try next?
>
> If this is not the correct forum for this kind of question, I'd like a
> referral if you know.
>
> Thanks for any consideration.
>
> Herb, WB1DSW.NH.USA.NOAM
> whftherb@nh.ultranet.com

36. Re: [HFSIG:2518] Infra-red MD problem

From: k5cnf1@juno.com (Richard B Morrow)
Date: Mon, 13 Oct 1997 15:54:04 -0500
To: hfsig@tapr.org
Message-ID: <19971013.155741.9406.1.k5cnf1@juno.com>
Rendered body type: text/plain

Bypass the ac line at the detector with .0001 caps.
73, K5CNF

On Mon, 13 Oct 1997 14:55:05 -0500 (CDT) Herb Salls WB1DSW
<whftherb@nh.ultranet.com> writes:
>Hello all.
>
>This doesn't really pertain to DSP or SS or much of the other esoteric
>topics I see here. Let me just pose this little situation I have with
>the hope that someone out there may have run across this before.
>
>I operate a full-time HF forwarding system (CLOVER-II) just north of
>the
>Boston area. At night, I switch the focus of the operation to 40
>meters.
>I have just recently installed infra-red motion detectors on the
>outside
>lighting. Whenever the 40 meter system fires up, I have one MD that
>constantly triggers on. It stays on for the full four minutes (the
>threshold I have set) then extinguishes only to re-trigger when and if

>the 40 meter transmitter is keyed. The 40 meter system consists of a
>Kenwood transceiver, and an all-band dipole fed through a tuner.
>
>While I'm mildly worried about the electric bill, I'm more concerned
>about how this looks to the folks who live around me. I've tried
>moving
>the antenna away from it, though I can't say that it's any great
>distance (less than a foot). I've tried changing the the MD and even
>the brand of MD. I've tried another HF transciever. I've tried
>calling
>the MD manufacturer (Stanley Works) - needless to say they weren't too
>sympathetic. I've run out of simple fixes that I can think of.
>
>Do we have any other "great thinkers" out there who might drop me a
>hint
>as to what to try next?
>
>If this is not the correct forum for this kind of question, I'd like a
>referral if you know.
>
>Thanks for any consideration.
>
>Herb, WB1DSW.NH.USA.NOAM
>whftherb@nh.ultranet.com
>
>

37. Re: [HFSIG:2518] Infra-red MD problem

From: Mike Fahmie <MPFahmie@lbl.gov>
Date: Mon, 13 Oct 1997 14:58:00 -0700
To: hfsig@tapr.org
Message-ID: <199710132159.OAA01600@mh1.lbl.gov>
Rendered body type: text/plain

At 02:55 PM 10/13/97 -0500, you wrote:

>I have just recently installed infra-red motion detectors on the outside
>lighting. Whenever the 40 meter system fires up, I have one MD that
>constantly triggers on. It stays on for the full four minutes (the
>threshold I have set) then extinguishes only to re-trigger when and if
>the 40 meter transmitter is keyed. The 40 meter system consists of a
>Kenwood transceiver, and an all-band dipole fed through a tuner.
>

Unless your harmonics are REALLY bad (infra-red!), the problem is probably
being caused by RF pickup on the 110 AC line to the motion detectors. Try
.001uF bypasses both across the line and to ground, passing both power leads
through a ferrite choke wouldn't hurt either. The line should see the choke
first. Be sure to use high voltage capacitors, 600 V or higher.

(Take two ferrites and call me in the morning!)

-Mike-

Mike Fahmie   -   Advanced Light Source
Lawrence Berkeley National Laboratory
(510) 486-4030   -   MPFahmie@lbl.gov
WA6ZTY @ N6EEG

38. Re: Error Condition Re: unsubscribe

From: jodie hoffman <jhoffma@tombstonepl.lib.az.us>
Date: Tue, 14 Oct 1997 08:35:20
To: hfsig@tapr.org
Message-ID: <3.0.1.32.19971014083520.008fca50@tombstonepl.lib.az.us>
Rendered body type: text/plain

Thank you, I did in fact wish to unscribe from all of your tapr and aprs
file servers.

Richard L. Homer

At 05:43 PM 10/13/97 -0500, you wrote:
>
>We are sorry, but this system sensed the following request which may have
been
>inadvertedly sent to this list:
>
>UNSUBSCRIBE
>

>If your posting was intentional, please accept our apologies and resend your
>mail message, making sure you do not include anything that may look like a
>request in the first line of the body of the actual message. If this was
>indeed a request please resend it to listproc@tapr.org
>Your entire message
>is copied below.
>
>--

>unsubscribe
>
>Richard Homer @ jhoffma@tombstonepl.lib.az.us
>
>
>

39. Re: [HFSIG:2521] Re: Infra-red MD problem

From: David A Willmore <willmore@cig.mot.com>
Date: Tue, 14 Oct 1997 14:53:20 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <9710141453.ZM26362@whelk>
Rendered body type: text/plain

On Oct 13, 5:05pm, Mike Fahmie wrote:
> Subject: [HFSIG:2521] Re: Infra-red MD problem
> At 02:55 PM 10/13/97 -0500, you wrote:
>
> >I have just recently installed infra-red motion detectors on the outside
> >lighting. Whenever the 40 meter system fires up, I have one MD that
> >constantly triggers on. It stays on for the full four minutes (the
> >threshold I have set) then extinguishes only to re-trigger when and if
> >the 40 meter transmitter is keyed. The 40 meter system consists of a
> >Kenwood transceiver, and an all-band dipole fed through a tuner.
> >
>
> Unless your harmonics are REALLY bad (infra-red!), the problem is probably
> being caused by RF pickup on the 110 AC line to the motion detectors. Try
> .001uF bypasses both across the line and to ground, passing both power leads
> through a ferrite choke wouldn't hurt either. The line should see the choke
> first. Be sure to use high voltage capacitors, 600 V or higher.
>
> (Take two ferrites and call me in the morning!)

If you use the cap solution, make sure you don't use ceramics! Try to get some
AC power line approved 'self-healing' power filter caps. If you can't find
any, send me your address and I'll send you a few.

When ceramics go bad, they die short. Not something you want on your AC line!
 They make a nice little explosion. Mylar and paper-metal caps fail by arcing
across a small section of the dielectric which fries the conductive places on
both sides--making a gap, not a short.

Cheers,
David N0YMV

40. New viterbi decoder release

From: Phil Karn <karn@ka9q.ampr.org>
Date: Tue, 28 Oct 1997 01:25:35 -0800 (PST)
To: hfsig@tapr.org
Cc: karn@homer.ka9q.ampr.org
Message-ID: <199710280925.BAA05316@homer.ka9q.ampr.org>
Rendered body type: text/plain

I have released a new version (2.1) of my Viterbi Decoder software
package.

This version includes an encoder and decoder for the k=7 rate=1/3 NASA
standard code, in addition to the existing k=7 r=1/2 code. Both block
and stream mode versions of the encoders and decoders are provided.

It also includes a general purpose block mode Viterbi decoder written
to handle any arbitrary code rate (1/N) and polynomial. (Because this
one was written for flexibility, it is slower on the standard k=7
r=1/3 and 1/2 codes than the dedicated routines, which were written
for speed.) The general purpose decoder includes a list of polynomials
of various lengths to play with, including the r=1/6 k=15 code used as
the inner code on Mars Pathfinder. (It's a very strong code, but boy
is it slow - only 386 bps on a 166 MHz Ultra Sparc 1!)

There is also a new, experimental Perl script that generates optimized
C code for an encoder and Viterbi decoder for an arbitrary set of
polynomials.

The URL is

http://people.qualcomm.com/karn/feccode/viterbi2.1.tar.gz

or

http://people.qualcomm.com/karn/feccode/viterbi21.zip

The archives include a readme file.

--Phil

November 1997

Original source file: hfsig.9711
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9711

1. TCM

```
From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 04 Nov 1997 20:43:43 +0000
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <345F88FF.94C1BA27@dircon.co.uk>
Rendered body type: text/plain
```

Hello All,

Things have been a bit quiet recently!

While surfin I found this URL that has an interesting Tutorial
on TCM (Trellis Code Modulation). All the rage in the DSTO parallel
tone modems.

http://lynx.neu.edu/home/httpd/e/ebrodeur/tcm/index.html

-Charles G4GUO

---

2. WARNING WARNING TREAT AS TRUTH UNTIL OTHERWISE PROVEN!!

```
From: kd4tqz@juno.com (Lee H. Dahlen)
Date: Tue, 11 Nov 1997 20:28:49 -0500
To: ehorton@juno.com, hfsig@tapr.org, KE4YWO@aol.com, ad4li@juno.com,
        JHais28374@aol.com, Robert_East@compuserve.com, b1blancr@gte.net,
        Laserdan@loclnet.com, robwilliams@InfoAve.Net, dahleng@shelby.net,
        norahd@juno.com, kf4fmc@juno.com, andrew_williams@shop.juno.com,
        w4cw@blue-ridge.net, n4xcv@juno.com, gregs@abts.net, n4woe@juno.com,
        postoffice@staff.juno.com, ARuppe4849@aol.com, w4cw@grits.net,
        ke4arz@juno.com, puppy@bellsouth.net, paul@spiritual-discoveries.com,
        ke4uey@bellsouth.net, kf4bwd@preferred.com, kf4icn@juno.com,
        kc4opb@shelby.net, kf4skc@juno.com, kf4ked@juno.com, KF4EZR@aol.com,
        JNewhous@kentelec.com, dhsessom@juno.com, bbrock@i-america.net,
        VFC1801@aol.com, billybobfisher@juno.com, jrs@conninc.com
Message-ID: <19971111.202856.14398.0.kd4tqz@juno.com>
Rendered body type: text/plain
```

THIS WAS FORWARDED TO ME AND I WILL TREAT IT AS FACT UNTIL PROVED
OTHERWISE! THIS WILL PROBABLY AFFECT THOSE OF YOU THAT BANK ONLINE TRADE
STOCKS ETC.   BE CAREFUL!! LEE

I'll try and cut through the crap, and try to get to the point of
this letter. I used to work for America Online, and would like to
remain anonymous for that reason. I was laid off in early September,
but I know exactly why I was laid off, which I will now explain:

Since last December, I had been one of the many people assigned to
design AOL 4.0 for Windows (AOL 4.0 beta, codenamed Casablanca). In
the beginning, I was very proud of this task, until I found out the
true cost of it. Things were going fine until about mid-February, when
me and 2 of my colleagues started to suspect a problem, an unexplainable
'Privacy Invasion', with the new version. One of them, who is a
master programmer, copied the finished portion of the new version (Then
'Build 52'), and took it home, and we spent nearly 2 weeks of sleepless
nights examining and debugging the program, flipping it inside-out, and
here is what we found.

Unlike all previous versions of America Online, version 4.0 puts
something in your hard drive called a 'cookie'. (AOL members click
A HREF="aol://4344:1047.g334.8411481.532897009">here for a
definition). However, the cookie we found on Version 4.0 was far
more treacherous than the simple internet cookie. How would you like
somebody looking at your entire hard drive, snooping through any
(yes,any) piece of information on your hard drive. It could also read
your password and log in information and store it deep in the program
code.

Well, all previous versions, whether you like it or not, have done
this to a certain extent, but only with files you downloaded. As me and
my colleagues discovered, with the new version, anytime you are signed
on to AOL, any top aol executive, any aol worker, who has been sworn to
secrecy regarding this feature, can go into your hard drive and
retrieve any piece of information that they so desire. Billing,
download records, e-mail, directories, personal documents, programs,

financial information, scanned images, etc ... Better start keeping all
those pictures on a floppy disk!

This is a totally disgusting violation of our rights, and your right
to know as well. Since this is undoubtably 'Top Secret' information
that I am revealing, my life at AOL is pretty much over. After
discovering this information, we started to inform a few other workers
at America Online, so that we could get a large enough crew to stop this
from happening to the millions of unfortunate and unsuspecting America
Online members. This was in early August. One month later, all three
of us were unemployed. We got together, and figured there was something
we had to do to let the public know.

Unemployed, with one of us going through a divorce (me) and another
who is about to undergo treatment for Cancer, our combined financial
situation is not currently enough to release any sort or article. We
attepted to create a web page on three different servers containing
in-depth information on AOL 4.0, but all three were taken down within
2 days. We were running very low on time (4.0 is released early this
winter), so we figured our last hope to reveal this madness before it
effects the people was starting something similar to a chain letter,
this letter you are reading. Please do the following, to help us
expose AOL for who they really are, and to help us and yourself recieve
personal gratification for taking a stand for our freedom:

1. Forward this letter to as many people as you can (not just
friends and family, as many as you can!)
2. Tell people who aren't on America Online in person, especially
important people (Private Investigators, Government workers, City
Council)
3. If the information about the new version isn't exposed by the
time aol is released early this winter, for your own protection, DON'T
DOWNLOAD AOL 4.0 UNDER ANY CONDITION !!!

Thank you for reading and examining this information. Me and my
colleagues hope that you will help us do the right thing in this
situation. Enjoy America Online (just kidding!).

------=_NextPart_000_01BCEB0F.9AAE9220--

--------------9BF2D06E057489C18D5A1616--

--------- End forwarded message ----------
--------- End forwarded message ----------

3. Re: [HFSIG:2526] WARNING WARNING

From: Antony Martin <anthony.martin@gecm.com>
Date: Wed, 12 Nov 1997 17:57:38 +0000
To: hfsig@tapr.org
Message-ID: <971112175738.12809@oas01.gmcl.co.uk>
Rendered body type: text/plain

The previous posting is a hoax e-mail. Its widely reported on the net as
such.

     AOL email hoax spreads superstition about cookies

A chain email is circulating the preposterous story that America Online has developed a cookie that can
send
files from a user's computer to the company, reported ZD News and CNET. Our warning page explains that
this
can't be done with cookies (but it can be done with Java on several bug-ridden versions of the two major
browsers).

Of course, any company whose software you run could instruct your computer to send anyone any
information it
has access to. Unless you have the source code and compile the program yourself, you have no way of
finding
out whether it might do this. You are trusting every company whose software you run on your PC with all
the
information on it. If you use AOL, you are already giving them a complete history of everything you look
 at on
the net, which goes into their databases. For this reason many people concerned about their privacy
choose
small ISPs that don't sell behavioral data about them to marketers.

A snippet from
http://www.junkbusters.com/ht/en/new.html

DO NOT email it to anyone else. Email hoaxes have become more prevalent
and faster spreading than viruses.

Anthony.

4. Updated EVM interface for Coherent BPSK

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 14 Nov 1997 10:14:06 -0800
To: DSP@tapr.org, HFSIG@tapr.org
Message-ID: <199711141814.KAA20143@peak.org>
Rendered body type: text/plain

Hi folks,

Pretty quiet these days - guess everyone's out having fun?

Just an update on some new code options for those interested in doing BPSK
on HF. I have added an option to my EVM interface program for use with the
VE2IQ, Coherent, program. That allows one to use a "new" waveform called the
"BLOB" waveform. The code package includes source code containing several
compile-time switches that accomodate all the various waveforms that
Coherent supports. The default is the blob waveform that is used as of lately.

Please look for COHEVM20.ZIP in ftp.tapr.org/tapr/SIG/dsp/recent_uploads.
It should appear there as soon as Tim has had the opportunity to move it
there (thanks Tim!).

As this is rather experimental stuff, I would appreciate feedback. The code
have been tested to work quite nicely during my regular skeds on BPSK with
Bill, WB6RIJ in Santa Barbara, but we would very much welcome any others
that would like to join in.

Have fun,

--Johan, KC7WW

5. BPSK skeds

From: "Donald E. Haselwood" <k4jpj@appstate.campus.mci.net>
Date: Fri, 14 Nov 1997 22:42:17 -0500
To: hfsig@tapr.org
Message-ID: <199711150242.VAA28193@aus-f.mp.campus.mci.net>
Rendered body type: text/plain

Is anyone doing skeds, or running a beacon, on BPSK nowadays? Sometime back
I listened, but was unable to hear any sigs (I think it was 14.082 Mhz or
something like that--nice RTTY sig there!). The propagation seems to be
improving so the maybe some longer paths might just be receivable.

73's
Don, W4DH

6. Re: [HFSIG:2529] BPSK skeds

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 14 Nov 1997 19:47:17 -0800
To: hfsig@tapr.org
Message-ID: <199711150347.TAA26498@peak.org>
Rendered body type: text/plain

>Is anyone doing skeds, or running a beacon, on BPSK nowadays? Sometime back
>I listened, but was unable to hear any sigs (I think it was 14.082 Mhz or
>something like that--nice RTTY sig there!). The propagation seems to be
>improving so the maybe some longer paths might just be receivable.
>
>73's
>Don, W4DH
>
>

WB6RIJ and myself is on 7081 (carrier) almost every evening around 5:00 -
5:30 PM PST. Earlier this summer there were regular cross-country QSO's
going on the same frequency.

If you are ready to roll, let us know and we can arrange for a sked.

73

Johan

7. Phase Invariant Convolutional Codes

From: Charles Brain <chbrain@dircon.co.uk>
Date: Mon, 17 Nov 1997 12:48:44 +0000
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <34703D2C.1DF26D7D@dircon.co.uk>
Rendered body type: text/plain

Hello all,

I wonder if someone can help me?

I am working on implementing a QPSK modem with convolutional
encoding for an H.F modem.

I have been looking at Phase Invariant Convolutional Codes,
this is where you differential encode a bit stream, convolutionally
encode the result, then map it to you transmitted signal.
On receive you do the reverse. The convolutional code I am
looking at using is a 180 deg phase-invariant or transparant code.

For an H.F modem is it better to use this class of code
or to do the differential operation after coding? At the moment
I do the differential encoding last, but I am using
a BPSK scheme, I am about to change it to QPSK with soft decisions.

My grey cells seem to tell me that the latter is best, the
book I am reading is principally about wireline modems not HF and
I can see why he would want to that with them.

- Charles G4GUO

8. Re: [HFSIG:2531] Phase Invariant Convolutional Codes

From: RLANIER@mailb.harris.com (RLANIER)
Date: Tue, 18 Nov 1997 12:52:37 -0500
To: hfsig@tapr.org
Message-ID: <471d5170@mailb.harris.com>
Rendered body type: text/plain

Charles,

     Where can I get more information on this technique? I did a search on
     the web, but came up empty.

     Tony KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:2531] Phase Invariant Convolutional Codes
Author: hfsig@tapr.org at smtp
Date: 11/17/97 6:57 AM

Hello all,

I wonder if someone can help me?

I am working on implementing a QPSK modem with convolutional
encoding for an H.F modem.

I have been looking at Phase Invariant Convolutional Codes,
this is where you differential encode a bit stream, convolutionally
encode the result, then map it to you transmitted signal.
On receive you do the reverse. The convolutional code I am
looking at using is a 180 deg phase-invariant or transparant code.

For an H.F modem is it better to use this class of code
or to do the differential operation after coding? At the moment
I do the differential encoding last, but I am using
a BPSK scheme, I am about to change it to QPSK with soft decisions.

My grey cells seem to tell me that the latter is best, the
book I am reading is principally about wireline modems not HF and
I can see why he would want to that with them.

- Charles G4GUO

9. Re: [HFSIG:2532] Re: Phase Invariant Convolutional Codes

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 18 Nov 1997 23:05:14 +0000
To: hfsig@tapr.org
Message-ID: <34721F2A.284819D@dircon.co.uk>
Rendered body type: text/plain

RLANIER wrote:
>
>   Charles,
>
>   Where can I get more information on this technique? I did a search on
>   the web, but came up empty.
>
>   Tony KE4ATO
>
> _____________________________ Reply Separator _____________________________
Hello Tony,

The book I have is "Convolutional Coding, Fundamentals ans Applications"
by L.H Charles Lee
Pub Artech House, ISBN 089006-914X

I think I have sorted out what is going on, the Covolutional code is one
that when all the input bits are inverted it still decodes, only the
output is
just inverted.

The book describes Multi-dimentional Trellis Codes and how to combine
decoding and channel equalisation. Plus appendices that list optimum
codes
for certain things, like optimum mother codes for puncturing etc.

- Charles G4GUO

10. Further-updated EVM interface for Coherent BPSK

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 20 Nov 1997 08:42:17 -0800
To: DSP@tapr.org, HFSIG@tapr.org
Message-ID: <199711201642.IAA06112@peak.org>
Rendered body type: text/plain

Hi Folks,

Another update on code for those interested in doing BPSK on HF using the
VE2IQ, "Coherent" program in conjunction with a DSP56002EVM. Better than 1
ppm timing accuracy is now possible (thanks to G3PLX for the hint).

The results are available as COHEVM21.ZIP on
ftp.tapr.org/tapr/SIG/dsp/recent_uploads. Please the included commented
source code on how to go about this.

A further bit of good news; since this application does not rely on the
accuracy of the CPU clock any longer, you may retain the ceramic resonator
part, if you have one of those.

Enjoy!

--Johan, KC7WW

PS. (Tim, could you move the code to the recent-upload directory - thanks much!)

11. COHEVM23 available

From: forrerj@peak.org (Johan Forrer)
Date: Sun, 30 Nov 1997 15:58:21 -0800
To: HFSIG@tapr.org, DSP@tapr.org
Message-ID: <199711302358.PAA06413@peak.org>
Rendered body type: text/plain

Hi,

For those following the C-BPSK experimentation on HF:

I have uploaded COHEVM version 2.3 to the recent_uploads in the DSP file
area. This version has some minor improvements, mostly being "DL friendly".

If you are using the N1OWU BIOS2 loader in Flash, this is good news.

Look for COHEVM23.ZIP in ftp://ftp.tapr.org/tapr/SIG/dsp/recent_uploads as
soon as Tim has moved it (thanks Tim!).

Enjoy!

--Johan

December 1997

Original source file: hfsig.9712
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9712

1. C-BPSK Questions

From: "Wheatley, Maurice" <mwheatle@trsystems.com>
Date: Mon, 1 Dec 1997 10:44:42 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <0EBC18A2F84FD11185760000F80163EA5634@www.trsystems.com>
Rendered body type: text/plain

Hi.

Just started digging into the C-BPSK stuff and wrote a simple DSP-93
program to emulate the sigma-delta and SDdac interface used by Coherent
in the SPORTO mode. It all seems to work OK in loopback duplex mode but
haven't heard any action at least not on 14081. I have a few questions
regarding this mode.

1. SPORTO data coming out of the PC's serial port is 2's compl. 8 bit
data offset by 0x80. What is the format of the data going into the PC's
serial port? Looking at Johan's EVM program indicates it is 8 bit 2's
compl. differential data with no offset. Is this true?

2. What are the frequency accuracy requirements on the transmit/receive
frequencies as well as the 7200 Hz sample/bit rate clock? Some of the
notes talk of +-1 Hz accuracy. Is this resolution, drift, or absolute
accuracy? It seems a ppm spec would be helpful. <1ppm on at least my
humble xceiver is gonna be impossible.

3. What is the bandwidth of the "blob" waveform using the MS25 speed?

4. What is the proper operating procedures using this mode? How does
one ID etc.

Appreciate any feedback on all or parts of these questions.

Thanks 73's

Moe Wheatley AE4JY
mwheatle@trsystems.com

2. Re: [HFSIG:2536] C-BPSK Questions

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 01 Dec 1997 09:17:23 -0800
To: hfsig@tapr.org
Message-ID: <199712011718.JAA15201@peak.org>
Rendered body type: text/plain

Hi Moe,

For definitive answers, how about asking Bill de Carle, VE2IQ, the author of
C-BPSK (bill@ietc.ca). He has been extremely helpful in getting me
straightened out on the EVM and will love hearing from you.
The SD A/D format that I use in COHEVM was provided by Bill. I believe the
SPORT0 SD D/A format should be the same, however, I have not implemented
that part as I am doing wave generation and shaping myself.

>2. What are the frequency accuracy requirements on the transmit/receive
>frequencies as well as the 7200 Hz sample/bit rate clock? Some of the
>notes talk of +-1 Hz accuracy. Is this resolution, drift, or absolute
>accuracy? It seems a ppm spec would be helpful. <1ppm on at least my
>humble xceiver is gonna be impossible.

It is important not to let things get totally out of hand and strive to
attain short term 7200Hz accuracy of 1-2ppm. Using a dithered clock scheme
is OK as Coherent buffers UART input.

C-BPSK is very fussy about frequency. Consider that 10Hz frequency error is
equal to 180 degree phase shift, that means that every symbol would be in
error. 5Hz frequency error is right at the limit, but there is little that
can be done to track that kind of error. Personally, I have found that you
need to be, and remain, within 2 Hz. That goes without saying that a
transciever with 10Hz synth steps are very hard to work with - 1 Hz
resolution is what you want. Typically, you also want a rig that uses only
one reference source for deriving timing, and that preferably being a

temperature-stabilized reference oscillator.

>3. What is the bandwidth of the "blob" waveform using the MS25 speed?

Measurement here shows BW of about 80Hz@-10dB, 100Hz@-20dB, 150Hz@-30dB, and
not much energy beyond that. This, incidentally, is for the real SD
hardware. Beware that the BLOB format duty cycle is low and you may need to
include additional amplification to get reasonable output power.

>4. What is the proper operating procedures using this mode? How does
>one ID etc.

Bill has some documentation in his Coherent program package that is worth
looking at. But most is by seat of the pants.
A few hints: Start each over with a string of right brackets "}}}}}}", this
helps others to synchronise - very useful in round table QSOs. Your ear
will get trained quickly as to which ET mode is being used - it used to be
quite distinct with the older waveforms, but needs a bit getting used to
with the BLOB format. We do manual CW-ID as there is no automatic CW-ID.

Easiest is to hook up with a C-BPSK station and learn how it works on the
air. Coherent has a very complicated cockpit, however, be assured that most
is for informational purposes only. Lately, there has been activity on
18.081000 MHz, and WB6RIJ and I have regular skeds on 7.081000 MHz at 5:00 -
5:30 PST. You are welcome to join us. Also check out the BPSK reflector for
announcements.

Hope this helps, 73

--Johan

3. BPSK demo

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 05 Dec 1997 09:43:40 -0800
To: HFSIG@tapr.org
Message-ID: <199712051744.JAA12439@peak.org>
Rendered body type: text/plain

Hi all,

For those interested in a demo of how HF BPSK performs under really adverse
conditions, download and unzip the file obtainable from:

ftp.tapr.org/tapr/SIG/dsp/recent_uploads/pskdemo.zip

This package contains a short sound clip of the signal. It is an example of
C-BPSK after running through my SHARC-based HF channel simulator. Test
conditions are set for CCIR "POOR" with SNR at -10dB. The accompanying text
file contains the results for various forward error-correcting schemes
currently in use in C-BPSK (Coherent 7.2) and PSK31.

73,

--Johan, KC7WW
.

4. Re: [HFSIG:2538] HFSIG digest 442

From: CDrentea <CDrentea@aol.com>
Date: Sun, 7 Dec 1997 21:06:19 EST
To: hfsig@tapr.org
Message-ID: <be310265.348b5626@aol.com>
Rendered body type: text/plain

In a message dated 97-12-06 12:28:43 EST, you write:

<< This package contains a short sound clip of the signal. It is an example
of
 C-BPSK after running through my SHARC-based HF channel simulator. Test
 conditions are set for CCIR "POOR" with SNR at -10dB. The accompanying text
 file contains the results for various forward error-correcting schemes
 currently in use in C-BPSK (Coherent 7.2) and PSK31.
  >>
About this, I dont know.

5. 40 meter antenna

From: RLANIER@mailb.harris.com (RLANIER)
Date: Sat, 13 Dec 1997 12:33:36 -0500

To: hfsig@tapr.org
Message-ID: <492c6100@mailb.harris.com>
Rendered body type: text/plain

Hi all,

I would like to install a 40 meter antenna in a limited space. I live in an apartment and the only logical place to put one would be on the balcony (my wife wouldn't care to have an antenna strung across the apartment, not to mention that it would be unsafe). I measured the balcony and it is (HxWxD) 9' x 13' x 6' approx. This isn't enough room for a full-sized or even a quarter wave antenna.

Can I uses a eighth wave or 12th wave antenna with adequate results? Has anyone had a similar experience or know of someone who did? I would really appreciate your help.

BTW, what is everyone's opinion on the new ARRL antenna book?

Tony KE4ATO

6. Re: [HFSIG:2540] 40 meter antenna

From: k5cnf1@juno.com (Richard B Morrow)
Date: Sun, 14 Dec 1997 10:07:11 -0600
To: hfsig@tapr.org
Message-ID: <19971214.100712.6510.1.k5cnf1@juno.com>
Rendered body type: text/plain

Check out http://www.antennex.com

On Sat, 13 Dec 1997 11:36:08 -0600 (CST) RLANIER@mailb.harris.com
(RLANIER) writes:
>   Hi all,
>
>   I would like to install a 40 meter antenna in a limited space. I
>live
>   in an apartment and the only logical place to put one would be on
>the
>   balcony (my wife wouldn't care to have an antenna strung across
>the
>   apartment, not to mention that it would be unsafe). I measured
>the
>   balcony and it is (HxWxD) 9' x 13' x 6' approx. This isn't enough
>room
>   for a full-sized or even a quarter wave antenna.
>
>   Can I uses a eighth wave or 12th wave antenna with adequate
>results?
>   Has anyone had a similar experience or know of someone who did? I
>
>   would really appreciate your help.
>
>   BTW, what is everyone's opinion on the new ARRL antenna book?
>
>   Tony KE4ATO
>
>

7. Offline Simulator

From: "Wheatley, Maurice" <mwheatle@trsystems.com>
Date: Mon, 15 Dec 1997 10:05:10 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <0EBC18A2F84FD11185760000F80163EA563E@www.trsystems.com>
Rendered body type: text/plain

Does anyone know of a program available that implements an HF simulator using the Watterson model that takes as input a .wav file of the input signal and outputs to another .wav file the simulated results? There are real time programs available running on various platforms but was wondering if an offline version was floating about.

Thanks,

Moe Wheatley AE4JY
mwheatle@trsystems.com

8. Re: [HFSIG:2542] Offline Simulator

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 15 Dec 1997 08:35:04 -0800
To: hfsig@tapr.org
Message-ID: <199712151635.IAA17865@peak.org>
Rendered body type: text/plain

Moe,

>
>Does anyone know of a program available that implements an HF simulator
>using the Watterson model that takes as input a .wav file of the input
>signal and outputs to another .wav file the simulated results? There are
>real time programs available running on various platforms but was
>wondering if an offline version was floating about.

For what it's worth:

I do not know of a simulator that works off .WAV files, and wonder of how
much value that will be.
Consider that typical HF conditions have fade rates in the order of
fractions of a Herz or perhaps as high as 10 Hz in exreme cases; you would
need an exceptionally large .WAV file (i.e. of sufficient durarion) to
produce any kind of usable statistics. Gathering statistics on a short
sample recoding where you perhaps had one or two fades, is stretching things
a bit. Repeatedly playing the sample over, unfortunately only produces
statistics of that bit of sample characteristics - that may not be a
representative situation. Actually, if you have any kind of ECC involving
time interleaving, you will need sufficient time to allow the processing of
several transmitted blocks of data.

Of course, if you could dedicate a hard drive (cheap nowadays) to making a
recoding lasting a few minutes that may become usable. For testing slow rate
BPSK systems, it would not be unusual to play a message of duration say 5-10
minutes. Other high speed HF protocols may require a sample of duration,
anything from 50 seconds to several minutes for a typical message.

My 2 cents worth.

--Johan, KC7WW

9. Simulator idea

From: "Wheatley, Maurice" <mwheatle@trsystems.com>
Date: Mon, 15 Dec 1997 14:06:19 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <0EBC18A2F84FD11185760000F80163EA563F@www.trsystems.com>
Rendered body type: text/plain

Hi Johan,
Yep I agree that long duration messages are needed. Memory is dirt
cheap now with CD-R blanks under 2 bucks and the recorders the same
price as small hard drives. I was thinking of creating long ie. up to
an hour's worth of CD quality audio(44.1Khz 16 bit 2 channel) of various
signal conditions. The CD audio data rate is around 10.58Mbytes/min so
an hour is < 650 Megs.(almost as big as a typical Microsoft app ;>))

Playback on a regular CD audio player could then be used to stimulate
the system under test freeing up a separate PC or DSP board just for
generating test signals. Perhaps each track could contain a different
channel condition, the second stereo channel a noise free version for
comparison, etc.

With 2GB and greater HDrives becoming the norm, wav files(over 3 hours
worth) could also be stored online and played back over a sound card,
however that eats up another PC just for generating signals.

All the above is within my reach, the channel simulator is the missing
piece. That was where I was coming from. Thanks for the response.
Hope to be on CBPSK soon. The final piece is figuring a way to make my
xceiver tune in <10Hz increments or be absolute in accuracy within 1 Hz.

73's

Moe Wheatley AE4JY
mwheatle@trsystems.com

>
>
>Of course, if you could dedicate a hard drive (cheap nowadays) to
making a
>recoding lasting a few minutes that may become usable. For testing slow

rate
>BPSK systems, it would not be unusual to play a message of duration say
5-10
>minutes. Other high speed HF protocols may require a sample of
duration,
>anything from 50 seconds to several minutes for a typical message.
>
>My 2 cents worth.
>
>--Johan, KC7WW

10. Re: [HFSIG:2544] Simulator idea

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 15 Dec 1997 15:44:13 -0800
To: hfsig@tapr.org
Message-ID: <199712152344.PAA29575@peak.org>
Rendered body type: text/plain

Hi Moe,

Seems you have a number of good suggestions - amazing how quickly technology
has changed and makes one rethink things?

>All the above is within my reach, the channel simulator is the missing
>piece. That was where I was coming from. Thanks for the response.

You may want to retrieve and old publication by Ehrman, Bates, Escle and
Kates on "Real-time software simulation of the HF Radio channel" in IEEE
Transactions on Communications, Vol. COM-30, no.8, of August 1982, pages
1809-1817. It includes valuable references. Should not be too difficult to
implement for someone with your background and capabilties.

>Hope to be on CBPSK soon. The final piece is figuring a way to make my
>xceiver tune in <10Hz increments or be absolute in accuracy within 1 Hz.

I have a similar predicament with my old ICOM 751. What I am planning on
doing is to tap the 455 KHz IF of the 751 and externally downconverting that
to around 12 KHz I/Q and then implementing a dual-conversion IF in DSP with
a digital "BFO" before the Coherent interface. For that, we have a
high-performance mixer in the works and Bill, WB6RIJ, is busy helping
fabricate it. The plan is to use a standard EVM for this DC receiver - might
be the start of a number of other related projects like DGPS etc... but more
about that later.

We are certainly looking forward hearing you on BPSK - we have been enjoying
excellent results under all conditions.
Incidentally, we also recently have had good results with the 15-tone
parallel tone AX.25-based modem of Pawel named "NEWQPSK." This was on 40
meters and we managed to download good-sized files over the 1500 mile path
between WB6RIJ and myself. The opportunities are mind boggling :-)

73, let me know if you need further help.

--Johan

11. Re: [HFSIG:2543] Re: Offline Simulator

From: Antony Martin <anthony.martin@gecm.com>
Date: Tue, 16 Dec 1997 13:51:11 +0000
To: hfsig@tapr.org
Message-ID: <971216135111.25831@oas01.gmcl.co.uk>
Rendered body type: text/plain

> Moe,
>
> >
> >Does anyone know of a program available that implements an HF simulator
> >using the Watterson model that takes as input a .wav file of the input
> >signal and outputs to another .wav file the simulated results? There are
> >real time programs available running on various platforms but was
> >wondering if an offline version was floating about.
>
> For what it's worth:
>
> I do not know of a simulator that works off .WAV files, and wonder of how
> much value that will be.
> Consider that typical HF conditions have fade rates in the order of
> fractions of a Herz or perhaps as high as 10 Hz in exreme cases; you would
> need an exceptionally large .WAV file (i.e. of sufficient durarion) to
> produce any kind of usable statistics. Gathering statistics on a short

> sample recoding where you perhaps had one or two fades, is stretching things

I've used sampled audio modem recordings for testing modem BER vs SNR.
I used files of up to 10 minutes to test raw BERs up to 1e-5.
Stored as 8-bit Mu-law 8kHz that's a 4.8Meg file.

Considering 2 Gig HDD are now standard, that's small beer.

I'm on a Unix machine; YMMV (rest of the world: YkmMV). Other software
products seem to assume large files don't exist.

- ---aside
Whereas 8-bit mu-law 8.000kHz RIFF .wav files can be readily created using
audio file format conversion programs, PC audio players only seem to
support an unknown subset of the RIFF file format. 8-bit and 16-bit
linear seem to work OK. RIFF format specifies host-byte-ordering
rather than some fixed byte ordering, making them non-portable, but
this seems to be ignored and many programs assume Intel/Micro$oft
byte ordering. (RIFF .WAV != Microsoft .WAV???)

I haven't got a channel sim sourcecode (yet) but I'd like one.

I spoke to Alexander Kurpiers (DL8AAU) at HF97 and he said he had
Ptolemy watterson models. I suggested that the Ptolemy code generation
facilities could be used to produce stand-alone channel simulator
programs, either for DSP or as general C sourcecode.
I haven't followed this up... perhaps I should?

Anthony.

- --
A.N. Martin   anthony.martin@gecm.com
Snr. Development Engineer
GEC-Marconi Communications Ltd.
These may or may not be the opinions of my employer.
- --

12. Vertically-polarized vs. horizontally-polarized antennae

From: RLANIER@mailb.harris.com (RLANIER)
Date: Tue, 16 Dec 1997 13:56:45 -0500
To: hfsig@tapr.org
Message-ID: <496cda20@mailb.harris.com>
Rendered body type: text/plain

Hi all,

     I was wondering, what are the consequences of using a
     vertically-polarized antenna to hear horizontally-polarized signals?
     I've read that the difference in polarization can be quite drastic in
     the receive quality. Also, aren't most vertically-polarized antennae
     omni-directional, further making them more susceptible to
     interference?

     I'm trying to make a decision as to which antenna configuration I want
     to use for 40 meters and I need to weigh all the factors before I make
     a purchase.

     Thanks in advance,

     Tony KE4ATO

13. Re: [HFSIG:2547] Vertically-polarized vs. horizontally-polarized antennae

From: k5cnf1@juno.com (Richard B Morrow)
Date: Tue, 16 Dec 1997 15:56:22 -0600
To: hfsig@tapr.org
Message-ID: <19971216.161408.3294.0.k5cnf1@juno.com>
Rendered body type: text/plain

If you can build a small transmitting loop, you will have all bases
covered. I am talking about a 10 ft diameter loop. It will also work on
75. If you are interested, let me know.
73, K5CNF

On Tue, 16 Dec 1997 12:59:41 -0600 (CST) RLANIER@mailb.harris.com
(RLANIER) writes:
>   Hi all,
>
>   I was wondering, what are the consequences of using a
>   vertically-polarized antenna to hear horizontally-polarized

>signals?
>   I've read that the difference in polarization can be quite
>drastic in
>   the receive quality. Also, aren't most vertically-polarized
>antennae
>   omni-directional, further making them more susceptible to
>   interference?
>
>   I'm trying to make a decision as to which antenna configuration I
>want
>   to use for 40 meters and I need to weigh all the factors before I
>make
>   a purchase.
>
>   Thanks in advance,
>
>   Tony KE4ATO
>
>

14. Re: Offline Simulator

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Wed, 17 Dec 1997 10:57:25 +0100
To: hfsig@tapr.org
Message-ID: <3497A205.181A@ife.ee.ethz.ch>
Rendered body type: text/plain

Antony Martin wrote:

> I haven't got a channel sim sourcecode (yet) but I'd like one.
>
> I spoke to Alexander Kurpiers (DL8AAU) at HF97 and he said he had
> Ptolemy watterson models. I suggested that the Ptolemy code generation
> facilities could be used to produce stand-alone channel simulator
> programs, either for DSP or as general C sourcecode.
> I haven't followed this up... perhaps I should?

In my Linux HF protocol stuff I have what I understand to
be a Watterson channel simulator (not necessarily correct :-))
The random number generator might not be good enough
for accurate simulations, though - and it depends on the
quality of the random number generator in the C library.
It should be fairly easy to adapt to PCM ("wav") I/O.

Tom

http://www.ife.ee.ethz.ch/~sailer/ham/linux/hf.html

15. Re: [HFSIG:2548] Re: Vertically-polarized vs.

From: RLANIER@mailb.harris.com (RLANIER)
Date: Wed, 17 Dec 1997 09:00:05 -0500
To: hfsig@tapr.org
Message-ID: <497d97a0@mailb.harris.com>
Rendered body type: text/plain

I am interested in this transmitting loop you discussed. Where can I
     get more information?

     Tony   KE4ATO

_____ Reply Separator _____
Subject: [HFSIG:2548] Re: Vertically-polarized vs.
Author: hfsig@tapr.org at smtp
Date: 12/16/97 4:22 PM

If you can build a small transmitting loop, you will have all bases
covered. I am talking about a 10 ft diameter loop. It will also work on
75. If you are interested, let me know.
73, K5CNF

On Tue, 16 Dec 1997 12:59:41 -0600 (CST) RLANIER@mailb.harris.com
(RLANIER) writes:
>   Hi all,
>
>   I was wondering, what are the consequences of using a
>   vertically-polarized antenna to hear horizontally-polarized
>signals?

>   I've read that the difference in polarization can be quite
>drastic in
>   the receive quality. Also, aren't most vertically-polarized
>antennae
>   omni-directional, further making them more susceptible to
>   interference?
>
>   I'm trying to make a decision as to which antenna configuration I
>want
>   to use for 40 meters and I need to weigh all the factors before I
>make
>   a purchase.
>
>   Thanks in advance,
>
>   Tony KE4ATO
>
>

16. Re: [HFSIG 2548:] HF Polarization

From: "Borcher, David (NG-EX)" <DBorcher@pioneer.nog.nlvl.com>
Date: Wed, 17 Dec 1997 07:25:57 -0800
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <CDFDEF962C55D1119FC3006097C510D0198034@nogalesxch.147.84.168.IN-ADDR.ARPA>
Rendered body type: text/plain

Hi Tony

There are some loop antennas in the ARRL Antenna Book, and they work.

I have also used the MFJ 1788 loop (40-15M) and it works too though
it is physically much smaller than some of those in the Antenna Book.
But it works over a lot of bands without external tuners.

BTW, much I have seen says that the polarization of signals received
after a trip thru the ionosphere is essentially random (and constantly
changing) which means it probably doesn't matter what YOU use, other
than the vertical pattern of your antenna of choice.

Vertical MONOPOLES are pretty much omni-directional, but ANY antenna
(yagi, log-periodic, curtain, quad, to name a few) can be constructed to
have
vertical polarization, and their directionality isn't affected.

Dave K7NG

17. Re: [HFSIG:2550] Re: Vertically-polarized vs.

From: Kok Chen <kchen@apple.com>
Date: Wed, 17 Dec 97 09:17:15 -0800
To: <hfsig@tapr.org>, <RLANIER@mailb.harris.com>
Message-ID: <199712171716.JAA11616@scv3.apple.com>
Rendered body type: text/plain

>   I am interested in this transmitting loop you discussed. Where can I
>   get more information?
>
>   Tony   KE4ATO

Try back issues of Communications Quarterly. There were
articles on small loops perhaps two years back.

If you have access to the Web, articles in

  http://www.antennex.com/

may be of interest to you. This is the online version of
the AntenneX periodical that suddenly went defunct a couple
of years ago.

It is by subscription, but there are many articles, under
"Preview Articles" that are freely accessible. Look for
articles on Loops and "Magnetic" antennae if that is what you
are interested in. Ted Hart, writing in AntenneX back then,
helped popularise small loops. For a nice tutorial about
small loops, go to

  http://ourworld.compuserve.com/homepages/csl/magloop.htm

Another Web page that I highly recommend are the pages of

L. B. Cebik, W4RNL,

  http://funnelweb.utcc.utk.edu/~cebik/radio.html

Cebik has an article on antenna forms other than the simple
vertical that have vertical polarisation. There is a whole
family of them. So, antennas can be directional and yet
be vertically polarised. However, if your intended use
is HF (for DX), it doesn't matter much if the transmitter
antenna polarisation matches receiver's. (My experiece
with ground wave propagation on HF, is it does matter.
But _only_ ground wave.)

Personally, I would stay away from the small magnetic loops
if you have any kind of space at all. But that is just
MHO. And, not to worry about polarisation if you are
working HF (this is _HF_SIG, isn't it? :-) :-).)

73

Kok Chen, AA6TY

18. Re: Vertically-polarized vs.

From: k5cnf1@juno.com (Richard B Morrow)
Date: Wed, 17 Dec 1997 18:48:49 -0600
To: hfsig@tapr.org
Message-ID: <19971217.194210.3718.0.k5cnf1@juno.com>
Rendered body type: text/plain

On Wed, 17 Dec 1997 12:44:31 -0600 (CST) Kok Chen <kchen@apple.com>
writes:
>>   I am interested in this transmitting loop you discussed. Where
>can I
>>   get more information?
>>
>>   Tony   KE4ATO

> http://www.antennex.com/
>
>may be of interest to you. This is the online version of
>the AntenneX periodical that suddenly went defunct a couple
>of years ago.
>
>It is by subscription, but there are many articles, under
>"Preview Articles" that are freely accessible. Look for
>articles on Loops and "Magnetic" antennae if that is what you
>are interested in. Ted Hart, writing in AntenneX back then,
>helped popularise small loops. For a nice tutorial about
>small loops, go to
>
> http://ourworld.compuserve.com/homepages/csl/magloop.htm
>
>Another Web page that I highly recommend are the pages of
>L. B. Cebik, W4RNL,
>
> http://funnelweb.utcc.utk.edu/~cebik/radio.html
>
>Cebik has an article on antenna forms other than the simple
>vertical that have vertical polarisation. There is a whole
>family of them. So, antennas can be directional and yet
>be vertically polarised. However, if your intended use
>is HF (for DX), it doesn't matter much if the transmitter
>antenna polarisation matches receiver's. (My experiece
>with ground wave propagation on HF, is it does matter.
>But _only_ ground wave.)
>
>Personally, I would stay away from the small magnetic loops
>if you have any kind of space at all. But that is just
>MHO. And, not to worry about polarisation if you are
>working HF (this is _HF_SIG, isn't it? :-) :-).)

Any particular reason? Ben Smith, W4KSY, has all the room for phased 40
meter verticals, yet he built a loop and is very pleased with his loop
and thinks it is a great DX antenna.

73, K5CNF

19. Re: [HFSIG:2553] Re: Vertically-polarized vs.

From: Kok Chen <kchen@apple.com>
Date: Wed, 17 Dec 97 20:20:53 -0800
To: <hfsig@tapr.org>
Message-ID: <199712180420.UAA28672@scv2.apple.com>
Rendered body type: text/plain

>Any particular reason? Ben Smith, W4KSY, has all the room for phased 40
>meter verticals, yet he built a loop and is very pleased with his loop
>and thinks it is a great DX antenna.

A small transmitting loop (I assume we are talking about
diameter < .1*lambda) has no more gain than a dipole. But the
complexity of constructing it is much higher than a dipole.
Yes, you can readily orient it for vertical polarisation, which
was the original poster's query, but this is not a critical
parameter for HF.

The loop itself may not be complex to build, but the capacitor for
resonating the loop is another story. The AEA loop, for example, uses
humongous plates for the loop capacitor. And although there has been
ingenious constructions lately (e.g., copper pipes sliding into one
another,
separated by teflon material), mechanically tuning these antennas can
still be a pain, especially for a mechanically inept lid like me :-).

The Q has to held extremely high so the losses can be low (considering
the radiation resistance of one of these things are typically a fraction
of an ohm). So, you are constantly tuning the loop to resonance as you
QSY, neccessitating some kind of remote tuning. On 20m, we are talking
about retuning every 10 or 20 kHz. (Food for thought: low loss 160m
transmitting loop has barely enough bandwidth to pass an SSB signal! :-)

I have also had severe RFI problems when I used an AEA loop on my
apartment balcony; but your milage could vary. I have since stuck
to homebrew short verticals on my balcony.

But I think we are straying far away from HFSIG's intent :-).

73 es Happy Holidays,

Kok Chen, AA6TY

20. New version on PSK31 avalaible!!!

From: Eduardo Jacob <edu@kender.es>
Date: Thu, 18 Dec 1997 23:56:41 -0100
To: psk31@bipt106.bi.ehu.es
Message-ID: <3.0.5.32.19971218235641.007fb6c0@kender.es>
Rendered body type: text/plain

There is a new version of PSK31 available by Peter Martinez G3PLX.
Although it is compatible with older PSK31, it adds a QPSK mode with a
convolutional code (Viterbi). In the tests it seems to be more interesting
and usable in HF than previous version. In any case read the doc, as it is
a place to learn.

Several suggestions: At present, we are always beginning QSO calling CQ in
BPSK and once tuned going on QPSK. The tunning is critical and you will se
that a variation of 3.9 hz will convert the copy in unusable. There is now
a cross for the 4 phase changes.

The version is available at

ftp://det.bi.ehu.es/pub/ham/psk31/psk31evm.zip

You can also get it on a really new page almost empty at present devoted
to PSK31 and QPSK31 modes at:

http://bipt106.bi.ehu.es/psk31.html

Hopefully there will be soon a QPSK version for the Texas C50 DSK. We
would like also get ports to other hardware (analog devices 2181, or sound
blaster), or other Operating Systems....

Best regards

Eduardo/EA2BAJ

21. Re: Vertically-polarized vs.

From: k5cnf1@juno.com (Richard B Morrow)
Date: Thu, 18 Dec 1997 22:43:27 -0600
To: hfsig@tapr.org
Message-ID: <19971218.230124.10270.1.k5cnf1@juno.com>
Rendered body type: text/plain

On Wed, 17 Dec 1997 22:34:56 -0600 (CST) Kok Chen <kchen@apple.com>
writes:
>>Any particular reason? Ben Smith, W4KSY, has all the room for phased
>40 >>meter verticals, yet he built a loop and is very pleased with his
>loop >>and thinks it is a great DX antenna.
>
>A small transmitting loop (I assume we are talking about
>diameter < .1*lambda) has no more gain than a dipole. But the
>complexity of constructing it is much higher than a dipole.
>Yes, you can readily orient it for vertical polarization, which
>was the original poster's query, but this is not a critical
>parameter for HF.
>
If the loop is in the vertical plane, it radiates a signal that is
perpendicular to the conductor, therefore it radiates a signal that
covers from horizontal to vertical and all points in between. If it is
horizontal, it then radiates a vertically polarized signal in a circular
pattern.

>The loop itself may not be complex to build, but the capacitor for
>resonating the loop is another story. The AEA loop, for example, uses
>humongous plates for the loop capacitor. And although there has been
>ingenious constructions lately (e.g., copper pipes sliding into one
>another, >separated by teflon material), mechanically tuning these
antennas can >still be a pain, especially for a mechanically inept lid
like me :-).
>
Yes that is true, but that is why we are attempting to devise a better
way to tune the loop. The trombone method is the easiest way to do so at
this time, but there are some easier methods in the mill.

>The Q has to held extremely high so the losses can be low (considering
>
>the radiation resistance of one of these things are typically a
>fraction of an ohm). So, you are constantly tuning the loop to
resonance as you QSY, necessitating some kind of remote tuning. On 20m,
we are talking about retuning every 10 or 20 kHz. (Food for thought: low
loss 160m transmitting loop has barely enough bandwidth to pass an SSB
signal!
>:-)
>
Yes these comments are also true, but bandwidth enables the loop to be
very quiet on reception. The program that we use to calculate the loop
characteristics enables you to vary the parameters of the loop and the
input power as well, so as to determine what you can do with it. The size
of the loop will determine the bandwidth, and the loop should not be
larger than 1/8 wavelength circumference at the highest frequency you
intend to use it on, otherwise the Q becomes so low that it is hard to
feed with a loop coupling feed.

>I have also had severe RFI problems when I used an AEA loop on my
>apartment balcony; but your milage could vary. I have since stuck >to
homebrew short verticals on my balcony.
>
The reason you had so much RFI was due to the efficiency of the loop and
the fact that it coupled into the wiring in your apartment and loaded up
every semiconductor and other things that was in the near field of the
antenna. I don't know what power you were running, but less is better in
apartments anyhow to avoid getting lynched! Been there and been accused
of RFI. But I was innocent, the 500watt CB was the guilty one.

>But I think we are straying far away from HFSIG's intent :-).
>
Yes, but without an antenna HFSIG becomes irrelevant

>73 es Happy Holidays,
>
>Kok Chen, AA6TY
>
Same to you Kok,
K5CNF

22. New version on PSK31 avalaible!!!

From: Eduardo Jacob <edu@kender.es>
Date: Fri, 19 Dec 1997 20:23:40 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19971219202340.007d7800@kender.es>
Rendered body type: text/plain

Murphy's law has stroken once again. In the evening the network has begun not to work very well, and at present I think it is not working at all. I have asked and it is due to the update of several nodes of the network and I hope the service will be ready for sunday or monday (it doesn't depend on our instalations, it's on the campus side)

In any case, meanwhile, the file is available here, in another unrelated location:

http://www.kender.es/~edu/psk31evm.zip

Eduardo/EA2BAJ

23. Pactor I for the EVM

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 23 Dec 1997 09:27:13 -0800
To: HFSIG@tapr.org, DSP@tapr.org
Message-ID: <199712231728.JAA03133@peak.org>
Rendered body type: text/plain

Hi folks,

I uploaded my latest efforts for a full Pactor I implementation for the EVM. It is available at:

ftp://ftp.tapr.org/tapr/SIG/dsp/recent_uploads/evmpac1.zip

This is work-in-progress, but I have been testing it on the air with good results.

Season's Greetings to all, 73

--Johan

(Tim, could you be so kind to move the files please - thanks much!)

24. DSP56002EVM

From: Dennis Gabler <dennisg@microware.com>
Date: Tue, 23 Dec 1997 12:37:29 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <9712231837.AA01064@wahz>
Rendered body type: text/plain

Hello All,

My Mot distrubutor is out on holiday. I was curious as to what the DSP56002EVM boards are costing these days ?

Thanks,
Dennis

--
Dennis Gabler   --o--------------------------=o=--------------------------o--
                                              H
                                              H   http://www.qsl.net/w5dg
    W5DG/NNN0BQG/NNN0GAR-1____________________H   dennisg@microware.com
    Principal Software Engineer   Microware Systems Corp.

25. Re: [HFSIG:2559] DSP56002EVM

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 23 Dec 1997 12:01:17 -0800
To: hfsig@tapr.org
Message-ID: <199712232002.MAA24931@peak.org>
Rendered body type: text/plain

Dennis,

I got one for my 455KHz DSP IF strip project the other day; Was about $150 from Active/Future.

Johan

>
>Hello All,
>
>My Mot distrubutor is out on holiday. I was curious as to what the
>DSP56002EVM boards are costing these days ?
>
>Thanks,
>Dennis
>
>--
>Dennis Gabler   --o--------------------------=o=--------------------------o--
>   H
>   H   http://www.qsl.net/w5dg
>   W5DG/NNN0BQG/NNN0GAR-1___________________H   dennisg@microware.com
>   Principal Software Engineer   Microware Systems Corp.
>
>

26. Small bug on PSK31 3.0

From: Eduardo Jacob <edu@kender.es>
Date: Wed, 24 Dec 1997 00:41:40 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19971224004140.007ce3d0@kender.es>
Rendered body type: text/plain

The are problems at work with the email. The website and ftp site are
already working.

A note from Peter

There is a bug in PSK31 version 3.00 just released. If you select
MORSE mode while the AFC is set to ON and the DCD is on, then
the frequency spins off tune very quickly in one direction for ever. I
know the fix and will send it soon to the Website, but for the
moment, please make sure that AFC is OFF when you change
modes through the MORSE mode with the F1 key.

In version 3.01 I will also change the F1 key so that it just toggles
between BPSK and QPSK only, and the MORSE mode will be done
by alt-F6, because I do not think many people will use it anyway.

73
Peter G3PLX

27. Re: [HFSIG:2549] Re: Offline Simulator

From: Phil Karn <karn@qualcomm.com>
Date: Tue, 23 Dec 1997 19:17:07 -0800 (PST)
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <199712240317.TAA28495@servo.qualcomm.com>
Rendered body type: text/plain

>The random number generator might not be good enough
>for accurate simulations, though - and it depends on the
>quality of the random number generator in the C library.

A couple of my FEC packages (Fano and Viterbi) include routines for
generating high-quality Gaussian-distributed random numbers using
the "rejection method" described in Numerical Recipes.

The only real drawback is speed -- each pair of random numbers requires
a floating point log and a floating point square root, plus a couple
of divides.

Phil

28. Re: [HFSIG:2558] Pactor I for the EVM

From: Eduardo Jacob <edu@kender.es>
Date: Wed, 24 Dec 1997 17:54:02 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19971224175402.00823cc0@kender.es>
Rendered body type: text/plain

>I uploaded my latest efforts for a full Pactor I implementation for the EVM.
>It is available at:

        Thank you, but could it be possible send me a .CLD file without symbolic
information (plxboot loader doesn't work) and with the PTT on bit #4?

Best regards and thank you for your contribution.

Eduardo

PD: Have you tested QPSK mode of PSK31 on your channel simmulator, on the
air we can see on our roundtables clear improvements although tunning on an
established QPSK QSO is rather difficult.
PD2: Since I got your version of coherent I haven't seen any activity here.
The QSO that Bill had with G3IRM are now held to early for me (I am at work).
PD3: Have you tried the audio spectrum analyzer for the EVM of Peter?. It's
available on my ham page. He says he isn't interested promoting PLX
software...

my page (mainly on DSP (EVM) and ham)

http://det.bi.ehu.es/~jtpjatae/ham.html

Evmspec
ftp://det.bi.ehu.es/pub/ham/evm/Fuentes/evmspec.zip
The bootloader
ftp://det.bi.ehu.es/pub/ham/evm/Fuentes/plxboot.zip
PD4: Good christmas

29. Re: [HFSIG:2563] Re: Pactor I for the EVM

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 24 Dec 1997 12:18:26 -0800
To: hfsig@tapr.org
Message-ID: <199712242019.MAA10595@peak.org>
Rendered body type: text/plain

Hello Eduardo,

>>I uploaded my latest efforts for a full Pactor I implementation for the EVM.
>>It is available at:
>
>   Thank you, but could it be possible send me a .CLD file without symbolic
>information (plxboot loader doesn't work) and with the PTT on bit #4?
>
>   Best regards and thank you for your contribution.
>
>   Eduardo
>

Can I really recommend trying Doug's (N1OWU) version of DL - I found it
works just fine in loading those .LOD files.

PB4, as far as I know, is allocated to the "frequency steering function" of
radio port #2, so probably is not a good idea to use that one for PTT. That
would create a lot of confusion for others. I followed the convention as
used for almost all the TAPR/Alef Null modems, i.e., using PB0 for PTT. I
hope it would not be too inconvenient to change your PTT allocation.

Something else that I forgot to mention in the README --- you can run this
Pactor implementation in a Win95 DOS box. Unlike my previous versions of
similar software, this one is considered "well behaved" in a Win95 sense.
The EVM is doing all the hard and timing-crital work for you.

There will be a number of revisions and new versions of this program package
ready for release as I get to fixing bugs and other annoyances and continue
with developing.

>PD: Have you tested QPSK mode of PSK31 on your channel simmulator, on the
>air we can see on our roundtables clear improvements although tunning on an
>established QPSK QSO is rather difficult.

I am planning on characterizing QPSK31 on the simulator shortly and will
share the results when it's ready. Perhaps over the holidays?

>PD2: Since I got your version of coherent I haven't seen any activity here.
>The QSO that Bill had with G3IRM are now held to early for me (I am at work).

We have been using the C-BPSK application (COHEVM) for some time now. It has
been tested on the air in communicating with another EVMs also with the
genuine VE2IQ hardware. Actually, it is almost impossible to tell the
difference.

>PD3: Have you tried the audio spectrum analyzer for the EVM of Peter?. It's
>available on my ham page. He says he isn't interested promoting PLX
>software...

I found this piece of code very useful, however, have not yet tried the
most-recent version of EVMSPEC.

Over the past few days, have been revisiting HF FAX using the EVM and
JVFAX7.1 Seems to work nicely too, but getting a bit boring now as I'm not
much of a meteorologist myself! Have you tried that one?

73

--Johan

30. Re: [HFSIG:2563] Re: Pactor I for the EVM

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 24 Dec 1997 16:25:34 -0800
To: hfsig@tapr.org
Message-ID: <199712250026.QAA03429@peak.org>
Rendered body type: text/plain

Eduardo,

>PD: Have you tested QPSK mode of PSK31 on your channel simmulator, on the
>air we can see on our roundtables clear improvements although tunning on an
>established QPSK QSO is rather difficult.

I made an extract of the simulator test results for BPSK and QPSK.
The test was for a CCIR Poor path with SNR set at at 1 dB and -5 dB.

For folks that are not too familiar with this mode:
The G3PLX version of 31 baud HF PSK does DBPSK and DQPSK encoding. The DBPSK
mode uses a form of coding called VARICODE, while DQPSK uses a
constraint-length 5 convolutional encoder in the trransmitter and a Viterbi
decoder in the receiver.

Anyone is interested in the full test results, including a sound clip of
what the signal sounds like at -5 dB SNR, please e-mail me and I'll forward
the 200K result file.

73

--Johan
KC7WW

Test: G3PLX's PSK31 (Differential PSK and QPSK)

Date: December 1997.
Tester: Johan Forrer, KC7WW
Simulator setup: CCIR Poor, for 1 dB and -5 dB SNR.
SNR: 10log(RMS signal/RMS noise).
Noise: 3KHz Gaussian.

.....this is an extract of a comprehensive test .....

The Message:

The TUNER program - TUNER.COM
---

1. This is a tuning aid to help get a received tone exactly on 800.0 Hz.
It should accept COM2, COM3, COM4 command line parameters (default is COM1)
and report CLIPPING (audio signal too strong for the sigma-delta circuit).

Test: BPSK at 1 dB SNR

      The TUNmR progrtm - l sNER.r OM
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1. Tmis is a tuaing aid to heap get a received tone exadtly on O00.0 Hz.
I should accept COM2, COM3, COM4 cemand line pnram tes (defaue oisD OM1E
and report C.p rPING (audio siðal too strong for the sigma-d|ta circuit).

```
Test: QPSK at 1 dB SNR
------------------------------------------------------------------------

The TUNER program - TUNER.COM
-----------------------------

1.  This is a tuning aid to help getw received tone exactly on 800.0 Hr.
It should accept COMv, COM3, COM4 command line parameters (default is COM1)
and report CLIPPING (audio signal too strong for the sigma-delta circuit).


Test: BPSK at -5 dB SNR
------------------------------------------------------------------------

 _he TUyER probram - TUNIcetCOa
----ð-------i t- t--------l---

1.  6is is a-uning aid tt heap get    received 0ne extc&y on eð0.0  roi.
It shouid accept COM2o  COM3, CO l4cohmand lineqarnmeters ðefault is COM1)
and teport CLIP-ING (audio signal to strong fetethe sigma-delta circuitP.n


Test: QPSK at -5 dB SNR
------------------------------------------------------------------------

      wðThe TUNeee p@gram - ðsN.R.COM
----------------st-----------

1.  Thas is a ðning aid to hele ðeo a received tone ex  ctly on 800.0 r z.
It  ei0ld accept COMm, COM3, COM4 command line parameters (default is COM1)
and r ðrt CLIPPING (rsdio siMtal too ltrong for the sigma-delta circuit).
```

31. Re: [HFSIG:2564] Re: Pactor I for the EVM

From: Eduardo Jacob <edu@kender.es>
Date: Thu, 25 Dec 1997 09:37:50 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19971225093750.00805100@kender.es>
Rendered body type: text/plain

>Can I really recommend trying Doug's (N1OWU) version of DL - I found it
>works just fine in loading those .LOD files.

The advantage of PLXBOOT is that both evmspec and psk31 automagically load
the code when they are told to do so and there is a program (loadevm.exe)
that is included with plxboot that allows to load .cld directly.

>PB4, as far as I know, is allocated to the "frequency steering function" of
>radio port #2, so probably is not a good idea to use that one for PTT. That
>would create a lot of confusion for others. I followed the convention as
>used for almost all the TAPR/Alef Null modems, i.e., using PB0 for PTT. I
>hope it would not be too inconvenient to change your PTT allocation.

To tell you the truth I dont recall why do I have this allocation. Would
it be possible to have the "facto standard" for port allocation?

>>PD3: Have you tried the audio spectrum analyzer for the EVM of Peter?. It's
>>available on my ham page. He says he isn't interested promoting PLX
>>software...

It's very usefull to locate the signal before attenting to tune it. Even
more interesting if you wnat to get into a roundtable that is already going
and involves rigs that deviate in frecuency.

>Over the past few days, have been revisiting HF FAX using the EVM and
>JVFAX7.1 Seems to work nicely too, but getting a bit boring now as I'm not
>much of a meteorologist myself! Have you tried that one?

Yes, I played a little. The problem is that I don't have worked before
with others interfaces so I don't know how it compares with them. The hffax
normally has the letters of the maps to small to recognize them. What I
used was the FSKIFACE interface and that can also be configured to do SSTV.
About SSTV modes, I have received with good results, but I have never
trasmited, I don't recall, but I think some VIS tones are not implemented.
Jabi EA2ARU, has promised a long time ago an AMPOP emulator, that is a
carefully written program that lobotomize the EVM and convert it in a dumb
(and very compatible) AMPOP hardware as suggested in the jvfax and hamcomm
documentations. He told me that he wrote it in C, but he has now rewritten
it in asm, the problem is the serial port bit allocation.

Eduardo/EA2BAJ

32. Nominations Sought for TAPR Board of Directors

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Thu, 25 Dec 1997 18:17:14 -0600
To: "TAPR-BB list mailing"<tapr-bb@tapr.org>,
"HF SIG list mailing"<hfsig@tapr.org>,
" Spread Spectrum "<ss@tapr.org>, " TAPR/AMSAT DSP "<dsp@tapr.org>,
"APRS SIG list mailing"<aprssig@tapr.org>,
"NETSIG list mailing"<netsig@tapr.org>,
"BBS SIG list mailing"<bbssig@tapr.org>,
" tacgps "<tacgps@tapr.org>, " MIC-E Beta "<mic-e@tapr.org>
Message-ID: <v03130309b0c8a625a448@[206.138.226.73]>
Rendered body type: text/plain

Nominations Sought for TAPR Board of Directors
(as printed in 1997 PSR #68, p. 34)

Tucson Amateur Packet Radio is incorporated in the State of Arizona as a non-profit scientific and educational institution. It is recognized by the IRS as a 501(c)3 tax-exempt organization for these same purposes. TAPR is governed by a 9-member Board of Directors. Each member of the Board serves a three year term. Every year three positions are up for election.

Board members are expected to attend two board meetings held in conjunction with the Dayton Hamvention and the ARRL and TAPR Digital Communications Conference. They participate in the decision-making process and provide guidance to the officers. They receive no pay and must defray most of their own expenses to attend meetings. Board members should be prepared to be active in the continuing Board deliberations, which are conducted via the Internet. Active participation in TAPR activities by Board members is important to the furtherance of the objectives of TAPR. The officers of TAPR are elected by the members of the Board at the annual Board of Directors meeting.

The current members of the Board of Directors and their expiration dates of their terms are:

*	John Ackermann, AG9V	1998	Vice President
*	Jim Neely, WA5LHS	1998	Treasurer
*	Barry McLarnon, VE3JF	1998	
	Steve Bible, N7HPR	1999	
	Gary Hauge, N4CHV	1999	
	Bob Hansen, N2GDE	1999	PSR Editor
	Greg Jones, WD5IVD	2000	President
	John Koster, W9DDD	2000	
	Mel Whitten, K0PFX	2000	

Nominations are now open for seats expiring in March 1998 (marked with an asterisk).

To place a person in nomination, please remember that he or she must be a member of TAPR. Confirm that the individual is willing to have their name placed in nomination. Send that person's name (or your own if you wish to nominate yourself) along with your call and their call, telephone numbers, mailing address, and Internet address. The person nominated should submit a short biographical sketch to be published along with the ballot.

Nominations and biographical sketches should be submitted to the TAPR office no later than December 31st, 1997.

Ballots will be mailed with the next PSR. TAPR will again use an Internet ballot, so read your ballot carefully. Results will be announced on March 30th, 1998.

Responsibilities of a board member include:
1) Attendance at both board meetings each year.
2) Regular participation with the continuous session of the board (currently held over the Internet). Typically this requires a minimum of 3 hours a week, although sometimes much more is required during active board discussions.
3) Participation with TAPR projects as volunteered. Board members, while not required, are involved with various project management, ongoing organization and/or supervision/liaison positions. Active board participation with various projects make many of the most important projects and tasks possible. Board members are expected to take an active part in TAPR in some form.

All nominated members will be placed on the ballot and the highest vote

receivers will be placed in the open board positions. If elected, two
board meetings in 1998 will be held. One will be during Dayton Hamvention
and the other during the ARRL and TAPR Digital Communications Conference.
All directors shall serve for a term of three years.

Tucson Amateur Packet Radio
8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399
---
e-mail: tapr@tapr.org   web: <http://www.tapr.org/>
ftp: ftp.tapr.org
phone: 940-383-0000   fax: 940-565-2544
---

33. Re: [HFSIG:2559] DSP56002EVM

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Fri, 26 Dec 1997 12:29:24 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.3.96.971226122734.10428A-100000@murdrum>
Rendered body type: text/plain

On Tue, 23 Dec 1997, Dennis Gabler wrote:

> My Mot distrubutor is out on holiday. I was curious as to what the
> DSP56002EVM boards are costing these days ?

Hi Dennis,

Motorola MSRP (Manufacturer Suggested Retail Price) is $149.95.

if anyone knows of a dealer selling the EVms for anything less, I'm sure
there are folks who would be interested!

73,
Tim

34. Unproto paths!

From: kd4tqz@juno.com (Lee H. Dahlen)
Date: Sat, 27 Dec 1997 12:54:46 -0500
To: hfsig@tapr.org
Cc: n4xcv@grits.net
Message-ID: <19971227.125456.14982.0.kd4tqz@juno.com>
Rendered body type: text/plain

THIS IS NOT A GRIPE LETTER JUST A REQUEST!!

Hf users on the System please unite!! There are alot of HF
gateways out there! 16 on my screen this morning all over the country!
Now if we can just let the vhfer's see ya! Some are running unproto
paths, some are not!! It doesnt matter ro me what you use, but PLEASE
PLEASE add the word GATE into ur (HF) unproto somwhere. What this will do
is take your 30m signal and if I (or every other gate) hear it I will put
it over on the 2m side so my locals can see you. I have people come over
often and say " I'm not sseing that much activity, and that is the reason
why! If you are running a KAM with gateway you must ck a few parameters
here is an example of what works for me. I am not subcribed so email me
direct at kd4tqz@juno.com

KAM PLUS PARAMETERS FOR GATEWAY USE (USING BOTH SIDES.)
(THESE ARE MUSTS FOR THE KAM TO WORK RIGHT)

NUMNODES: AT LEAST 1 (0 WILL NOT ALLOW GATE TO FUNCTION)
INIT: TERM
MYG: GATE
KNXC:ON (YES THIS IS A CMD!) THIS IS WHAT ALLOWS THE CROSS GATE TO
OPEN.
MAXUSERS: AT LEAST 2

UNPROTO: THIS IS THE BARE MINIMUM THAT WILL WORK! (TYPE IT IN THIS WAY)

GATE/APRS VIA (local digi),WIDE,TRACE

REMEMBER THIS IS FOR KAM PLUS USERS ONLY!

73'S TO ALL FROM LEE / KD4TQZ

35. SORRY!!!

From: kd4tqz@juno.com (Lee H. Dahlen)
Date: Sat, 27 Dec 1997 14:49:22 -0500
To: hfsig@tapr.org
Message-ID: <19971227.145402.15134.0.kd4tqz@juno.com>
Rendered body type: text/plain

This message concerning unproto paths was inadvertantly sent to the wrong
 sig! It wads supposed to go to APRSSIG. Please accept my sincere
appollogies for the error!   Lee

Please dont reply the msg was rerouted!   Thx.

36. New version 3.01 of PSK31 available

From: Eduardo Jacob <edu@kender.es>
Date: Sun, 28 Dec 1997 23:28:51 -0100
To: psk31@bipt106.bi.ehu.es
Message-ID: <3.0.5.32.19971228232851.0080a730@kender.es>
Rendered body type: text/plain

There is a new version of psk31 available at:

ftp://det.bi.ehu.es/pub/ham/psk31/psk31evm301.zip

The PSK31 page is located at

http://aintel.bi.ehu.es/psk31.html

Changes introduced in version 3.01
----
1. Bugfix: When switching to (or through) Morse mode with the AFC on, the
   receive and transmit frequencies changed continuously. Now fixed.

2. Previously, aborting a CQ call or a FileSend operation did not take
   place until the send buffer emptied. Now these operations no-longer feed
   through the send buffer and can therefore be aborted more quickly.

3. The MORSE mode is now on the alt-F6 key rather than in the F1 key
   sequence

37. Re: [HFSIG:2571] New version 3.01 of PSK31 available

From: Joe & Bonita Sanada III <heyco@esslink.com>
Date: Sun, 28 Dec 1997 19:27:34 -0500
To: hfsig@tapr.org
Message-ID: <34A6EE76.91C3291B@esslink.com>
Rendered body type: text/plain

Hi Ed do you have a Home web page this is Joe KA1ZXN i am not sure if its your
web site checked or not thanks alot Joe KA1ZXN..

Eduardo Jacob wrote:

> There is a new version of psk31 available at:
>
> ftp://det.bi.ehu.es/pub/ham/psk31/psk31evm301.zip
>
> The PSK31 page is located at
>
> http://aintel.bi.ehu.es/psk31.html
>
> Changes introduced in version 3.01
> ----------------------------------
> 1. Bugfix: When switching to (or through) Morse mode with the AFC on, the
>   receive and transmit frequencies changed continuously. Now fixed.
>
> 2. Previously, aborting a CQ call or a FileSend operation did not take
>   place until the send buffer emptied. Now these operations no-longer feed
>   through the send buffer and can therefore be aborted more quickly.
>
> 3. The MORSE mode is now on the alt-F6 key rather than in the F1 key
>   sequence

38. James Clerk Maxwell made easy ?

From: zs6awk@global.co.za (Danie Brynard)
Date: Wed, 31 Dec 1997 07:04:54 +0200
To: hfsig@tapr.org
Message-ID: <199712310604.IAA02541@mail.global.co.za>
Rendered body type: text/plain

From: zs6awk@global.co.za (Danie Brynard)
Subject: James Clerk Maxwell reference ?

Hope these questions is within the scope of amateur HF signal processing:
(if not perhaps it can aid our understanding of HF signals)

****
1. Can somebody recommend a very practical book on Maxwell's equations for
timevarying fields please ?
****

It could also be a PC program/magazine article or ordinary article. I am
particularly interested in something that presents a lot of practical
applications and examples.

I have also been wondering why exactly does our antennas radiate an EM
signal ? My understanding is that this problem cannot be solved by Maxwell's
equations but rather by quantum physics. Does somebody have an explanation
or can suggest a reference for further reading ?

****
2. Why does an antenna radiate ? (ie why are the photons produced at the
exciting RF frequency)
****

Thanking You

Danie zs6awk

++
Danie Brynard   Email: zs6awk@global.co.za
PO Box 15133,Lyttelton,0140,South Africa
GRID LOC: KG44CE,   Tel/Fax: (2712) 6640330
++

39. Best Noise Reduction Algorythms?

From: Rob Frohne <frohro@wwc.edu>
Date: Wed, 31 Dec 1997 15:54:19 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199712312354.PAA15433@adams.wwc.edu>
Rendered body type: text/plain

Hi All,

I've been reading the reviews on the newest Timewave DSP (DSP599zx) and it
looks like the hardware they are using isn't as good as my Motorola DSP56002EVM.
However, they receive rave reviews from QST and others. They list as adaptive
the noise reduction filters. I'm wondering what they are using for them. The
CW, SSB and other digital modes are using FIR linear phase filters, and I assume
they are using an LMS adaptive algorythm for the heterodyne reduction as it is
listed as adaptive also. I am wondering what the opinions of the group are
as to what would be the best noise reduction filter algorythm for QRN. Is anyone
working on this for the DSP56002EVM?

Thanks,

Rob, KL7NA

--
                              Rob Frohne
        E. F. Cross School of Engineering, Walla Walla College
        Phone: (509) 527-2075   FAX: (509) 527-2867
        Internet e-mail: frohro@wwc.edu

40. Re: [HFSIG:2574] Best Noise Reduction Algorythms?

From: Jim Sanford <jsanfor@exis.net>
Date: Wed, 31 Dec 1997 19:34:13 -0500
To: hfsig@tapr.org
Message-ID: <34AAE485.8363E334@exis.net>
Rendered body type: text/plain

Rob:
I have a DSP-59, and a -599zx (use the 59 mobile) and like them both. Believe that

it is LMS, and works better than anything else I've listened to.

73, J im
wb4gcs@amsat.org

Rob Frohne wrote:

> Hi All,
>
> I've been reading the reviews on the newest Timewave DSP (DSP599zx) and it
> looks like the hardware they are using isn't as good as my Motorola DSP56002EVM.
> However, they receive rave reviews from QST and others. They list as adaptive
> the noise reduction filters. I'm wondering what they are using for them. The
> CW, SSB and other digital modes are using FIR linear phase filters, and I assume
> they are using an LMS adaptive algorythm for the heterodyne reduction as it is
> listed as adaptive also. I am wondering what the opinions of the group are
> as to what would be the best noise reduction filter algorythm for QRN. Is anyone
> working on this for the DSP56002EVM?
>
> Thanks,
>
> Rob, KL7NA
>
> __
>   Rob Frohne
>   E. F. Cross School of Engineering, Walla Walla College
>   Phone: (509) 527-2075   FAX: (509) 527-2867
>   Internet e-mail: frohro@wwc.edu

January 1998

Original source file: hfsig.9801
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9801

1. Re: [HFSIG:2575] HFSIG digest 460

From: "B.HAINEAULT" <bhaineau@cts.com>
Date: Thu, 1 Jan 1998 16:42:02 -0800
To: <hfsig@tapr.org>
Message-ID: <000e01bd1717$3f370920$d805a3cd@bhaineau.cts.com>
Rendered body type: text/plain

I had a JPS NIR-10, a Timewave DSP59+ and I now have their DSP599zx. I agree that it surpasses JPS' offerings. I am very satisfied with its capabilities... and I can even mention that it has way more features than I really need. I use the voice and data filters, the noise reduction and the 60Hz filter.

Do they have the "best" noise reduction algorithms ? Hard to say... except that my experience indicates that they are one of the best out there. JPS' unit is a close second... but the sheer flexibility of Timewave's model makes it a great value.

73 de Bruno Haineault, VE2EQ/W6
San Marcos, California

ICQ Pager #: 335336
ICQ Pager address: http://wwp.mirabilis.com/335336

-----Original Message-----
From: hfsig@tapr.org <hfsig@tapr.org>
To: hfsig@tapr.org <hfsig@tapr.org>
Date: Wednesday, 1997 December, 31 22:18
Subject: [HFSIG:2575] HFSIG digest 460

>   HFSIG Digest 460
>
>Topics covered in this issue include:
>
> 1) Best Noise Reduction Algorythms?
> by Rob Frohne <frohro@wwc.edu>
> 2) Re: Best Noise Reduction Algorythms?
> by Jim Sanford <jsanfor@exis.net>
>
>--
>
>Date: Wed, 31 Dec 1997 15:54:19 -0800 (PST)
>From: Rob Frohne <frohro@wwc.edu>
>To: hfsig@tapr.org
>Subject: Best Noise Reduction Algorythms?
>Message-ID: <199712312354.PAA15433@adams.wwc.edu>
>
>Hi All,
>
>I've been reading the reviews on the newest Timewave DSP (DSP599zx) and it
>looks like the hardware they are using isn't as good as my Motorola DSP56002EVM.
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>CW, SSB and other digital modes are using FIR linear phase filters, and I assume
>they are using an LMS adaptive algorythm for the heterodyne reduction as it is
>listed as adaptive also. I am wondering what the opinions of the group are
>as to what would be the best noise reduction filter algorythm for QRN. Is anyone
>working on this for the DSP56002EVM?
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>Thanks,
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>Rob, KL7NA
>
>__
> Rob Frohne
> E. F. Cross School of Engineering, Walla Walla College

> Phone: (509) 527-2075 FAX: (509) 527-2867
> Internet e-mail: frohro@wwc.edu
>
>
>------------------------------
>
>Date: Wed, 31 Dec 1997 19:34:13 -0500
>From: Jim Sanford <jsanfor@exis.net>
>To: hfsig@tapr.org
>Subject: Re: Best Noise Reduction Algorythms?
>Message-ID: <34AAE485.8363E334@exis.net>
>
>Rob:
>I have a DSP-59, and a -599zx (use the 59 mobile) and like them both.
Believe that
>it is LMS, and works better than anything else I've listened to.
>
>73, J im
>wb4gcs@amsat.org

2. Re: Best Noise Reduction Algorithm

From: Alberto Di Bene <75800340@it.ibm.com>
Date: Fri, 2 Jan 1998 11:21:21 +0000
To: <hfsig@tapr.org>
Message-ID: <506010001204661600000 2L062*@MHS>
Rendered body type: text/plain

Rob, indeed the LMS works well for this purpose, provided you choose correctly
the various parameters, like mu,
 decay speed, and others. I wrote such a program for the DSK 'C50 made by TI,
and use it routinely in my HAM
 station, with very good results. But there is another method, which I haven't
so far had the chance to test, and I think
 it is used in the commercial NIR unit. Conceptually it is quite simple : take
the FFT of the incoming signal, and set
 to zero all the bins which are below a given threshold, adjustable with a
potentiometer (followed by an ADC).
 Then do an IFFT and you are done. It is on my to-do list, though not at the
highest priority.

73 de Alberto, I2PHD   dibene@usa.net

3. Re: [HFSIG:2577] Re: Best Noise Reduction Algorithm

From: Rob Frohne <frohro@wwc.edu>
Date: Fri, 2 Jan 1998 09:38:58 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <199801021738.JAA17016@adams.wwc.edu>
Rendered body type: text/plain

Hi Alberto,

Thanks for the reply. I've got a C50 DSK sitting around here. I wonder
if I could try your setup out. I'm beginning to see that it is as you
say. You need to be able to adjust the parameters just right. Do you
have an external control to adjust them or do you re-assemble again
and again until you like it?

Thanks,

Rob
>
> Rob, indeed the LMS works well for this purpose, provided you choose co=
> rrectly
> the various parameters, like mu,
> decay speed, and others. I wrote such a program for the DSK 'C50 made =
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> highest priority.

>
> 73 de Alberto, I2PHD   dibene@usa.net
> =
>
>

4. Re: [HFSIG:2577] Re: Best Noise Reduction Algorithm

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Sat, 03 Jan 1998 07:15:21 -0500
To: hfsig@tapr.org
Message-ID: <34AE2BD7.39C12DA2@worldnet.att.net>
Rendered body type: text/plain

Alberto Di Bene wrote:

> Rob, indeed the LMS works well for this purpose, provided you choose correctly
> the various parameters, like mu,
> decay speed, and others. I wrote such a program for the DSK 'C50 made by TI,
> and use it routinely in my HAM
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> so far had the chance to test, and I think
> it is used in the commercial NIR unit. Conceptually it is quite simple : take
> the FFT of the incoming signal, and set
> to zero all the bins which are below a given threshold, adjustable with a
> potentiometer (followed by an ADC).
> Then do an IFFT and you are done. It is on my to-do list, though not at the
> highest priority.
>
> 73 de Alberto, I2PHD   dibene@usa.net

Alberto:

This is not an easy algorithm to get right. If you "zero" out a bin, this will
cause a you problems
when you IFFT unless you handle it very carefully. Just as sudden sharp changes
in the time
domain (a spike) causes an effect on EVERY bin in the FFT, so will a sudden sharp
change
in the frequency domain cause a major effect on every time waveform sample when
you IFFT.
You need to smooth out what it is you are doing and "zeroing out a bin" is almost
never the right
thing to do.

Bob
N4HY

5. Re: [HFSIG:2579] Re: Best Noise Reduction Algorithm

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Sat, 3 Jan 1998 11:44:41 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.3.96.980103113625.15272A-100000@murdrum>
Rendered body type: text/plain

On Sat, 3 Jan 1998, Robert McGwier wrote:

> You need to smooth out what it is you are doing and "zeroing out a bin" is almost
> never the right
> thing to do.

Hi Bob,

I've been thinking about this in regard to some of my experience, and I'm
not too sure.

Counter example:

The notch filter on your HF rig. It is a very narrow, very deep filter.
Depending on the size of your FFT, the stop band of the notch filter can
be one FFT bin.

The results of zeroing one bin is like a notch filter.

If you have a constant value in all the bins of your FFT, this is white
noise. If you drop them all to zero, the noise is removed. If you can
detect the threshold level of this constant value in the bins, you can
zero these bins and leave the bins that are above the threshold alone
(your desired signal).

I'm digging through stuff...At one time I was playing with a hiss
reduction algorithm that did exactly what is proposed in this thread. I
think I got it from a prof at NMSU though.

Also, I believe some of the hiss reduction algorithms such as Dobly use
this method.

True, zeroing bins in the FFT will have an effect in the time domain, but
perhaps it is the desired effect if you zero the correct bins. I can do
some Matlab simulations, and if anyone is intersted, I can put it up on a
web site.

73,
Tim

6. Re: Best Noise Reduction Algorithm

From: Alberto Di Bene <75800340@it.ibm.com>
Date: Sun, 4 Jan 1998 01:50:41 +0000
To: <hfsig@tapr.org>
Message-ID: <506010001207l244000002L042*@MHS>
Rendered body type: text/plain

Rob, my setup is composed of the TI DSK 'C50 connected via the serial port
to a PC running DOS. I wrote a driving program on the PC that sends to the DSK
all the commands to activate and adjust LMS, Automatic Notch, AGC, Low Pass
and High Pass filters.
The serial protocol on the DSK is synchronized with the sampling rate
(9600 samples/sec) so there is no disruption of audio when the DSK and the PC
exchange data.
Just let me the time to translate into English the menus on the PC and, if
there is
interest, I will upload to the TAPR ftp server the DSK and the PC code.

The drawback of my setup is that you need a PC powered on near the station,
and this sometimes can cause birdies and etherodynes, if shielding is not very
effective. I have under construction a board with a 87C752 microcontroller, that
will take the role of the PC, so to have a standalone box, complete with an LCD
display, potentiometers and switches.

Alberto di Bene, I2PHD dibene@usa.net

7. Re: Best Noise Reduction Algorithm

From: Alberto Di Bene <75800340@it.ibm.com>
Date: Sun, 4 Jan 1998 01:54:43 +0000
To: <hfsig@tapr.org>
Message-ID: <506010001207l285000002L052*@MHS>
Rendered body type: text/plain

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Alberto di Bene, I2PHD dibene@usa.net

8. Re: Best Noise Reduction Algorithm

From: Alberto Di Bene <75800340@it.ibm.com>
Date: Sun, 4 Jan 1998 01:55:26 +0000
To: <hfsig@tapr.org>
Message-ID: <506010001207l288000002L082*@MHS>
Rendered body type: text/plain

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Alberto di Bene, I2PHD dibene@usa.net

9. Re: Best Noise Reduction Algorithm

From: Alberto di Bene <dibene@usa.net>
Date: Sat, 03 Jan 1998 19:05:23
To: hfsig@tapr.org
Message-ID: <19980104020523.19937.qmail@www07.netaddress.usa.net>
Rendered body type: text/plain

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Alberto di Bene, I2PHD dibene@usa.net
_____
Get free e-mail and a permanent address at http://www.netaddress.com

10. Re: [HFSIG:2580] Re: Best Noise Reduction Algorithm

From: "Keith McElveen" <jkm@mclean.sparta.com>
Date: Mon, 5 Jan 1998 08:45:25 -0500
To: <hfsig@tapr.org>
Message-ID: <01bd19e0$2d44e360$0170b99d@Clemson>
Rendered body type: text/plain

Hi Tim,
I for one would find this very interesting. I've been looking for new (or
ignored old) algorithms to do hiss and pop noise reduction in the "real"
world - I have yet to run across any robust algorithms other than the
classical low pass, LMS, and spectral subtraction ones.

Keith

-----Original Message-----
From: Tim Baggett <tim@acca.nmsu.edu>
To: hfsig@tapr.org <hfsig@tapr.org>
Date: Saturday, January 03, 1998 2:13 PM
Subject: [HFSIG:2580] Re: Best Noise Reduction Algorithm

>On Sat, 3 Jan 1998, Robert McGwier wrote:
>
>> You need to smooth out what it is you are doing and "zeroing out a bin"
is almost
>> never the right
>> thing to do.
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>Hi Bob,
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>I've been thinking about this in regard to some of my experience, and I'm
>not too sure.
>

>Counter example:
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>Depending on the size of your FFT, the stop band of the notch filter can
>be one FFT bin.
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>I'm digging through stuff...At one time I was playing with a hiss
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>some Matlab simulations, and if anyone is intersted, I can put it up on a
>web site.
>
>73,
>Tim
>
>
>

11. Re: Maxwell's Equations Made Easy (!?!)

From: DSCHULTZ@SSSP.HST.NASA.GOV
Date: Mon, 5 Jan 1998 15:04:11 GMT
To: hfsig@tapr.org
Cc: zs6qwk@global.co.za, DSCHULTZ@SSSP.HST.NASA.GOV
Message-ID: <980105150411.1862@SSSP.HST.NASA.GOV>
Rendered body type: text/plain

Danie Brynard, zs6awk, asked:

>1. Can somebody recommend a very practical book on Maxwell's equations for
>timevarying fields please ?

"The Science of Radio" by Paul J. Nahin (ISBN 1-56396-347-7), published by the American Institute of Physics, is a delightful book that bridges the gap between the highly mathematical university textbooks, the non-mathematical hobbyist publications such as The Radio Amateur's Handbook, and the completly non-technical radio history books that cover politics and business but ignore the technical history of radio. Most engineering and science textbooks tend to ignore history entirely, but Nahin's book covers the mathematical and technical development of the early days of radio.

This book assumes that you have had first year university level calculus and physics courses, it is not "dumbed down" by any means. Some chapter titles are: "Pre-Radio History of Radio Waves", "Antennas As Launchers and Interceptors of Electromagnetic Waves", "Mathematics of AM Sidebands", "First Continous Wave and Hetrodyne Concept", "Birth of Electronics", "Radio Spectrum of a Spark Gap Transmitter", "Multiplying by Squaring and Filtering", "Multiplying by Sampling and Filtering", "Synchronous Demodulation and Its Problems", and "Analytic Signals and Single-Sideband Radio".

Everyone who loves radio and is not afraid of Fourier transforms should read this book. It includes a quote from Richard Feynman's "Lectures on Physics" that "ten thousand years from now there can be little doubt that the most significant event of the 19th century will be judged as Maxwell's discovery of the laws of electrodynamics. The American Civil War will pale into provincial insignificance in comparison...", and from Einstein "One ought to be ashamed to make use of the wonders of science embodied in a radio set, the while appreciating them as little as a cow appreciates the botanic marvels in the plants she munches".

>I have also been wondering why exactly does our antennas radiate an EM
>signal ? My understanding is that this problem cannot be solved by Maxwell's
>equations but rather by quantum physics. Does somebody have an explanation

>or can suggest a reference for further reading ?

Quantum mechanics is not needed to explain EM radiation, classical physics is adequate. Indeed, radio waves were "discovered" in the late 19th century as a mathematical consequence of Maxwell's equations. Hertz then set out to experimentally demonstrate the existence of the waves that had been predicted by Maxwell.

>2. Why does an antenna radiate ? (ie why are the photons produced at the >exciting RF frequency)

Radio engineers generally don't talk about photons. They live in a purely classical world of time-varying electric and magnetic fields. The reason for this is that radio "photons" are exceptionally low energy compared to photons of light or gamma rays.

The energy of a photon is given by E=h*f where f is the frequency of the radiation and h is Planck's constant. For visible light at 500 nanometers, f = 6E+14 Hz or 600,000 GHz. The energy of such a photon is around 2 electron volts, enough to excite atoms into higher energy states. The emission or absorption of a photon of that energy is a noticable event.

For a radio signal at 1 GHz, the photon energy is a few millionth's of an electron volt, a tiny amount of energy that is not measurable in any practical way. The random thermal motion of electrons in your receiver generates enough noise to completly swamp any measurment of individual photons.

As you know from your physics class we can describe visible light as either a particle or a wave. Depending on what type of experiment we want to do, we can choose to think about light as either a particle or a wave and we freely switch back and forth between the two theories as needed. You need quantum mechanics to deal with the particle theory but the wave theory is purely classical physics. For radio waves, we NEVER talk about particles, only waves. On the other end of the spectrum, for gamma rays whose photon energy can be millions of electron volts, we ALWAYS talk about photons and never describe gamma rays as waves.

Dan Schultz N8FGV

12. Re: [HFSIG:2585] Re: Best Noise Reduction Algorithm

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 05 Jan 1998 08:53:16 -0800
To: hfsig@tapr.org
Message-ID: <199801051654.IAA21036@peak.org>
Rendered body type: text/plain

Hello all,

My 2 cents worth on this interesting subject:

There are some code for the EVM that is worth revisiting ..... LMS (and autonotching), spectral subtraction, and correlation - all are effective for noise reduction in one way or another. Some work better for some modes than others.

If you are have an EVM available, look for QRMQRN.ASM (LMS, Wiener filters), FFT-CUT.ASM (spectral subtraction), and CORELFLT.ASM (long correlator.) Locate these by following the links on the TAPR web page to a collection of code that I (KC7WW) put together. With these working examples, you can explore the mechanics of the various algorithms discussed before, and expand on that if you wish. Source code is included of course.

My experience has been that if you are interested in CW, take a look at the long correlator methods. That produces very pleasing effects, and does not contain the "artificial" overly-processed side effects that one hear when using the LMS or spectral subtraction method. Also note that spectral subtraction takes careful thought otherwise it produces some really weird side-effects when things are not done right. Apparently the combination of LMS with spectral subtraction has some desireable affects - I think JPS calls this "peaking subtraction".

Effective spectral subtraction on speach requires the algorithm to locate voiced vs. unvoiced periods as one way to effective adjust dynamic thresholds. Doug Hall (who developed the code for the JPS boxes) wrote a nice article on how this works in Personal Engineering magazine (sorry I don't have the exact reference on hand), also Dave Hershberger (W9GR) had an article in QST a while back where this issue was discussed in some detail,

including some good references. Check these out.

In conclusion, I also need to add that I also have found there are some personal biases that certain individuals have when it comes to noise reduction. What seems to matter is how long a period the individual need to be subjected to it, or whether or not it is for listening for weak signals, also the type of noise that needs to be dealt with. Dealing with one kind of noise may adversely affect a weak signal, but my reduce listening stress - Obviously an interesting but rather complex subject.

Trust this is of interest.

--Johan, KC7WW

>Hi Tim,
>I for one would find this very interesting. I've been looking for new (or
>ignored old) algorithms to do hiss and pop noise reduction in the "real"
>world - I have yet to run across any robust algorithms other than the
>classical low pass, LMS, and spectral subtraction ones.
>
>Keith
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>-----Original Message-----
>From: Tim Baggett <tim@acca.nmsu.edu>
>To: hfsig@tapr.org <hfsig@tapr.org>
>Date: Saturday, January 03, 1998 2:13 PM
>Subject: [HFSIG:2580] Re: Best Noise Reduction Algorithm
>
>
>>On Sat, 3 Jan 1998, Robert McGwier wrote:
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>>> You need to smooth out what it is you are doing and "zeroing out a bin"
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>>I'm digging through stuff...At one time I was playing with a hiss
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>>web site.
>>
>>73,
>>Tim
>>
>>
>>
>
>

13. Re: Best Noise Reduction Algorithm

From: Alberto di Bene <dibene@usa.net>
Date: Mon, 05 Jan 1998 10:12:55
To: hfsig@tapr.org
Message-ID: <19980105171255.18202.qmail@www03.netaddress.usa.net>
Rendered body type: text/plain

Gentlemen,
sorry for the multiple, misformatted appends
caused by blues of Lotus Notes.

I have ready the English version of DSPBOX, complete with
some documentation, but it seems that I have problems with
ftp put. Being behind a firewall doesn't help either.
I have the ZIP file ready, about 49 kbytes. If some kind
soul is willing to receive it as a mail attachment, and then
transfer it to the TAPR ftp site, please let me know.

Thanks and 73,
Alberto, I2PHD dibene@usa.net

_____
Get free e-mail and a permanent address at http://www.netaddress.com

14. Re: [HFSIG:2586] Re: Maxwell's Equations Made Easy (!?!)

From: Jim Sanford <jsanfor@exis.net>
Date: Mon, 05 Jan 1998 17:47:39 -0500
To: hfsig@tapr.org
Message-ID: <34B1630B.8B032F2C@exis.net>
Rendered body type: text/plain

Dan:
GREAT post! Thanks for sharing it; just added a new book to the "gotta buy"
list.

73, Jim
wb4gcs@amsat.org

DSCHULTZ@SSSP.HST.NASA.GOV wrote:

> Danie Brynard, zs6awk, asked:
>
> >1. Can somebody recommend a very practical book on Maxwell's equations for
> >timevarying fields please ?
>
> "The Science of Radio" by Paul J. Nahin (ISBN 1-56396-347-7), published
> by the American Institute of Physics, is a delightful book that bridges
> the gap between the highly mathematical university textbooks, the
> non-mathematical hobbyist publications such as The Radio Amateur's
> Handbook, and the completly non-technical radio history books that cover
> politics and business but ignore the technical history of radio. Most
> engineering and science textbooks tend to ignore history entirely, but
> Nahin's book covers the mathematical and technical development of the
> early days of radio.
>
> This book assumes that you have had first year university level calculus
> and physics courses, it is not "dumbed down" by any means. Some chapter
> titles are: "Pre-Radio History of Radio Waves", "Antennas As Launchers
> and Interceptors of Electromagnetic Waves", "Mathematics of AM Sidebands",
> "First Continous Wave and Hetrodyne Concept", "Birth of Electronics",
> "Radio Spectrum of a Spark Gap Transmitter", "Multiplying by Squaring
> and Filtering", "Multiplying by Sampling and Filtering", "Synchronous
> Demodulation and Its Problems", and "Analytic Signals and Single-
> Sideband Radio".
>
> Everyone who loves radio and is not afraid of Fourier transforms should
> read this book. It includes a quote from Richard Feynman's "Lectures on
> Physics" that "ten thousand years from now there can be little doubt that
> the most significant event of the 19th century will be judged as Maxwell's
> discovery of the laws of electrodynamics. The American Civil War will
> pale into provincial insignificance in comparison...", and from Einstein
> "One ought to be ashamed to make use of the wonders of science embodied
> in a radio set, the while appreciating them as little as a cow appreciates
> the botanic marvels in the plants she munches".
>
> >I have also been wondering why exactly does our antennas radiate an EM
> >signal ? My understanding is that this problem cannot be solved by Maxwell's
> >equations but rather by quantum physics. Does somebody have an explanation
> >or can suggest a reference for further reading ?

>
> Quantum mechanics is not needed to explain EM radiation, classical physics
> is adequate. Indeed, radio waves were "discovered" in the late 19th century
> as a mathematical consequence of Maxwell's equations. Hertz then set out
> to experimentally demonstrate the existence of the waves that had been
> predicted by Maxwell.
>
> >2. Why does an antenna radiate ? (ie why are the photons produced at the
> >exciting RF frequency)
>
> Radio engineers generally don't talk about photons. They live in a purely
> classical world of time-varying electric and magnetic fields. The reason
> for this is that radio "photons" are exceptionally low energy compared to
> photons of light or gamma rays.
>
> The energy of a photon is given by E=h*f where f is the frequency of the
> radiation and h is Planck's constant. For visible light at 500 nanometers,
> f = 6E+14 Hz or 600,000 GHz. The energy of such a photon is around 2
> electron volts, enough to excite atoms into higher energy states. The
> emission or absorption of a photon of that energy is a noticable event.
>
> For a radio signal at 1 GHz, the photon energy is a few millionth's
> of an electron volt, a tiny amount of energy that is not measurable in
> any practical way. The random thermal motion of electrons in your
> receiver generates enough noise to completly swamp any measurment of
> individual photons.
>
> As you know from your physics class we can describe visible light as
> either a particle or a wave. Depending on what type of experiment we
> want to do, we can choose to think about light as either a particle
> or a wave and we freely switch back and forth between the two theories
> as needed. You need quantum mechanics to deal with the particle theory
> but the wave theory is purely classical physics. For radio waves, we
> NEVER talk about particles, only waves. On the other end of the
> spectrum, for gamma rays whose photon energy can be millions of
> electron volts, we ALWAYS talk about photons and never describe
> gamma rays as waves.
>
> Dan Schultz N8FGV

15. Re: [HFSIG:2585] Re: Best Noise Reduction Algorithm

From: Phil Karn <karn@qualcomm.com>
Date: Mon, 5 Jan 1998 12:35:11 -0800 (PST)
To: hfsig@tapr.org
Cc: karn@servo.qualcomm.com
Message-ID: <m0xpJEt-000HSYC@laptop.ka9q.ampr.org>
Rendered body type: text/plain

>I for one would find this very interesting. I've been looking for new (or
>ignored old) algorithms to do hiss and pop noise reduction in the "real"
>world - I have yet to run across any robust algorithms other than the
>classical low pass, LMS, and spectral subtraction ones.

Well, how about 1) a good low-rate digital voice encoder plus 2)
forward error correction plus 3) a good digital modem? That will certainly
reduce hiss and pop noise...

Seriously -- now that we have so much digital technology in our hands,
why do we insist on using it mainly to implement analog modulation
schemes that are inherently limited in performance by the mathematics
that describe them?

Phil

16. Re: [HFSIG:2590] Re: Best Noise Reduction Algorithm

From: Brian Beezley <k6sti@n2.net>
Date: Mon, 5 Jan 1998 16:44:12 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19980105164434.193fae4c@mail.n2.net>
Rendered body type: text/plain

>Well, how about 1) a good low-rate digital voice encoder plus 2)
>forward error correction plus 3) a good digital modem? That will certainly
>reduce hiss and pop noise...

But only above threshold. I think the ear plus brain and an analog voice
SSB signal still beat any digital scheme for weak-signal recovery. If
anyone believes otherwise and has an algorithm, contact me and I'll propose

a business arrangement. If it can be done, there is plenty of money to be made in the amateur radio market.

>Seriously -- now that we have so much digital technology in our hands,
>why do we insist on using it mainly to implement analog modulation
>schemes that are inherently limited in performance by the mathematics
>that describe them?
>
>Phil
>

Because there aren't any advanced-technology digital signals to decode. That can be changed, but it requires personal initiative.

Brian Beezley, K6STI
k6sti@n2.net

17. Re: [HFSIG:2590] Re: Best Noise Reduction Algorithm

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Mon, 5 Jan 1998 18:56:50 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.3.96.980105185423.7182C-100000@murdrum>
Rendered body type: text/plain

On Mon, 5 Jan 1998, Phil Karn wrote:

> Seriously -- now that we have so much digital technology in our hands,
> why do we insist on using it mainly to implement analog modulation
> schemes that are inherently limited in performance by the mathematics
> that describe them?

Because it costs a lot more than my oatmeal box crystal radio!

But seriously, I don't know of any digital voice with FEC being used on HF, and the Kenwood TS870 sitting on my desk does need and use noise reduction. I imagine there are many other HF rigs out there that need noise reduction as well.

Just my 2 cents...

Tim

18. Re: [HFSIG:2587] Re: Best Noise Reduction Algorithm

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Mon, 5 Jan 1998 22:16:02 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.3.96.980105220550.7182I-100000@murdrum>
Rendered body type: text/plain

Hi Johan,

On Mon, 5 Jan 1998, Johan Forrer wrote:
> If you are have an EVM available, look for QRMQRN.ASM (LMS, Wiener filters),
> FFT-CUT.ASM (spectral subtraction), and CORELFLT.ASM (long correlator.)
> Locate these by following the links on the TAPR web page to a collection of
> code that I (KC7WW) put together. With these working examples, you can
> explore the mechanics of the various algorithms discussed before, and expand
> on that if you wish. Source code is included of course.

Yes, I played with QRNQRM a few years ago, and I did get it to work. I was mostly just playing around with it and other applications along with it when I was learning the EVM at NMSU.

I'd like to add, that the Alef Nul group has some documentation that may be of interest for those interested in using LMS in noise and interference reduction:

http://www.tapr.org/dsp/dsp4/application.html

> Effective spectral subtraction on speach requires the algorithm to locate
> voiced vs. unvoiced periods as one way to effective adjust dynamic
> thresholds. Doug Hall (who developed the code for the JPS boxes) wrote a
> nice article on how this works in Personal Engineering magazine (sorry I

Yeah, since unvoiced sheech patterns have a somewhat broadband noise
characteristic to them, you can eiasly cancel it out and only end up with
the voiced segments I imagine.

I don't know if it was Doug Hall's series or not, but there has been an
excellent series of DSP articles in Personal Engineering. I haven't been
handed PE at work recently though (guess I should get my own
subscription!)

73,
Tim

19. Re: [HFSIG:2591] Re: Best Noise Reduction Algorithm

From: Antony Martin <anthony.martin@gecm.com>
Date: Tue, 06 Jan 1998 12:54:42 +0000
To: hfsig@tapr.org
Message-ID: <980106125442.24380@oas01.gmcl.co.uk>
Rendered body type: text/plain

Hello all;

> >Well, how about 1) a good low-rate digital voice encoder plus 2)
> >forward error correction plus 3) a good digital modem? That will certainly
> >reduce hiss and pop noise...

Have you heard the new stanag 4479 800b/s voice coder? It's really
awful, almost entirely unintelligible IMO. If you thought LPC-10 2400b/s
wasn't exactly broadcast quality, it's much worse. It's said to
have a DRT score of 87(%), but that just tells us how useless the
DRT score is. I jest not, it's like someone wearing a gas mask and gargling
sump oil talking in 'action replay' slow motion down a telephone.
If the customers want it, and it's a standard, we will have to do it, but
I think it's a waste of time, no-one's gonna use it when they've
got it.

Give me a higher coding rate, less FEC and a few more channel errors anyday.
Uncoded 2400b/s speech does get better on good channels, but 800b/s speech
with FEC sounds bad on all channels!!!

I believe there are better quality systems than LPC for 2400b/s now,
probably of the IMBE family.

> But only above threshold. I think the ear plus brain and an analog voice
> SSB signal still beat any digital scheme for weak-signal recovery.

Listen to the comparisons of coded speech and noisy analogue channels at:
http://www-mobile.ecs.soton.ac.uk/speech_codecs/voicedemo/
If you can't play .au files there's lots of free software to translate
audio formats out there just waiting to be compiled.

Large interleave error correction that deals with HF fading can't be
used on speech because of the delay is unacceptable. So there isn't much
that FEC can do for speech on HF.

I'm thinking store-and-forward coded speech voicemail may be a good way
to support speech on poor links? RAM is cheap these days...

> >Seriously -- now that we have so much digital technology in our hands,
> >why do we insist on using it mainly to implement analog modulation
> >schemes that are inherently limited in performance by the mathematics
> >that describe them?
> >
> >Phil

> Because there aren't any advanced-technology digital signals to decode.
> That can be changed, but it requires personal initiative.

Ah, but there are. They're usually encrypted so listening wouldn't be
very rewarding! Unless you have 10000 supercomputers with 10000
processors each.

If you're into digital DX you can pick up the ALE messages from the US
naval fleet; ALE is early 1980s (?) so is not 'advanced-technology'.

> Brian Beezley, K6STI
> k6sti@n2.net

But the ultimate application of digits results in an HF mobile-phone

system. Is that what you're looking for?

_____

A.N.Martin   E-mail anthony.martin@gecm.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
_____

20. DSPBOX program for noise reduction

From: Alberto di Bene <dibene@usa.net>
Date: Tue, 06 Jan 1998 07:11:23
To: hfsig@tapr.org
Message-ID: <19980106141123.12253.qmail@www07.netaddress.usa.net>
Rendered body type: text/plain

Thanks to Eduardo, EA2BAJ, the DSPBOX.ZIP file has been uploaded
to the 'upload' directory of the TAPR ftp site. Now it should be moved
to the right directory by the site manager (Greg may be ?)
It is a program meant for the DSK 'C50, and I seem to remember that there
should be a directory for this DSK somewhere.

In case someone is interested to details of the implementation
of the program, inside the ZIP file there is a TXT file with
some documentation, though far from exhaustive.

I would be interested in reports on how the program works for you.
I like it, but, then, I wrote it....

Alberto, I2PHD

_____
Get free e-mail and a permanent address at http://www.netaddress.com

21. Question on status of digital voice

From: gc@cen.com (Gary Chatters)
Date: Tue, 6 Jan 98 10:49:10 EST
To: hfsig@tapr.org
Message-ID: <9801061549.AA04394@cen.com>
Rendered body type: text/plain

Phil writes:

>
>Well, how about 1) a good low-rate digital voice encoder plus 2)
>forward error correction plus 3) a good digital modem? That will certainly
>reduce hiss and pop noise...
>
>Seriously -- now that we have so much digital technology in our hands,
>why do we insist on using it mainly to implement analog modulation
>schemes that are inherently limited in performance by the mathematics
>that describe them?
>

One of the local repeaters is setup so they can link to the Internet
and use Internet phone (not sure of exactly which system). They have
occasional sessions where they make connections so they can have QSOs
with random global locations.

The audio quality is not good. It is quite understandable, but also
quite harsh sounding and not pleasant to listen to. I'd rather listen
to an SSB signal with the hiss and pop.

Some questions: Is this representative of low rate voice encoding
that might also be used on HF? Are there other systems that might
be tolerable to use? What's the state of the art?
Consider this a general inquiry from someone who has no experience
in this area.

73,
Gary

22. Re: [HFSIG:2591] Re: Best Noise Reduction Algorithm

From: Arvydas Vaisnys <vaisnys@zorba.jpl.nasa.gov>
Date: Tue, 06 Jan 1998 10:55:22 -0800
To: hfsig@tapr.org
Message-ID: <3.0.32.19980106105521.006e52ec@zorba.jpl.nasa.gov>
Rendered body type: text/plain

At 06:45 PM 1/5/98 -0600, you wrote:
>
>>Well, how about 1) a good low-rate digital voice encoder plus 2)
>>forward error correction plus 3) a good digital modem? That will certainly
>>reduce hiss and pop noise...
>
>
>But only above threshold. I think the ear plus brain and an analog voice
>SSB signal still beat any digital scheme for weak-signal recovery. If
>anyone believes otherwise and has an algorithm, contact me and I'll propose
>a business arrangement. If it can be done, there is plenty of money to be
>made in the amateur radio market.
>
>
>>Seriously -- now that we have so much digital technology in our hands,
>>why do we insist on using it mainly to implement analog modulation
>>schemes that are inherently limited in performance by the mathematics
>>that describe them?
>>
>>Phil
>>
>
>
>Because there aren't any advanced-technology digital signals to decode.
>That can be changed, but it requires personal initiative.
>
>
>Brian Beezley, K6STI
>k6sti@n2.net
>
Initiative and funding. Actually there occasionally are advanced-technology
digital signals to decode. As part of a VOA/JPL digital HF development
effort, we ran a couple of tests last year and transmitted a variety of
signal formats between Delano CA and Washington DC. The signal set included
MPSK, for M=2 to 64 and 16QAM and 64QAM. Different combinations of FEC and
time interleaving were also tried. BPSK modulated with a PN code was used
to measure the delay spread (multipath components) of the channel. Some of
the higher order modulation formats were used to transmit compressed audio
or a known sequence for bit error testing. DSB AM was also transmitted for
comparison.

Our receiver is implemented in sofware on a PC and unfortunately does not
run fast enough to recover the signal in real time. We downconvert the
received signal to a low IF and record on a DAT for later processing.

The tests yielded some interesting propagation measurements and performance
results. We managed to transmit 16 kbps compressed audio, using 8PSK and a
small amount of FEC, over some severe channels. The DSB AM signal
transmitted under the same conditions was pretty noisy. The digital signal
at this rate is basically AM quality and distortion and noise free, until
the link crashes.

The results of this task have been written up in reports to the VOA and
will eventually be presented at various conferences. The first paper at a
general conference will be for PTC'98 next week. It describes the results
of the latest test. If there is interest, I could send it to your ftp site.
It's a 2 Mbyte word document. Tell me where and how.

Meantime a not so short summary of the trades involved in digital HF that
covers the three items mentioned at the top:

The three factors that need to be overcome in a digital HF system (for
audio) are: 1) limited bandwidth; 2) low SNR, due to fading and
interference; and 3) multipath.

1. Bandwidth. The HF broadcast channel is 10 kHz, which allows
approximately 8 k symbols/sec (baud) with pulse shaping. Vocoder state of
the art at present is such that approximately 16 kbps is needed to
reproduce AM quality audio. To get this rate through the channel requires a
modulation order >2, i.e. at least 8PSK to allow for FEC and other
overhead. Higher modulation orders will allow higher data rate, but at the
expense of higher SNR requirement, or more coding which in turn reduces the
data rate. This trade can be shown to be counter productive when going much
above 8PSK in a Gaussian channel and becomes even worse in a multipath
channel.

It also helps to have a vocoder which can tolerate a moderate bit error
rate, say 10E-3 or higher. There are also tricks that can be played such as
repeating voice frames when a short error burst occurs or just squelching

the vocoder output. This gets you into a trade of how much time
interleaving you want to use.

2. SNR. Again, the higher the modulation order the higher the SNR required.
Assuming you have a good equalizer and can operate in a multipath channel,
you will find that bit error rate curves reach a fixed floor. In other
words, after a certain point you can continue to increase transmitter power
and it does not help any more. The bit error floor is higher for higher
modulation orders. For example, if the error floor for a particular
modulation order, say 8PSK is 10E-2, then you must supply enough FEC to
bring the error rate down to 10E-3 which the vocoder will tolerate.

For the above example, a Reed Solomon (255, 223) code which can correct 16
bytes out of the 255 byte block is about right. 8PSK with RS (255, 223) is
one of the combinations we used in the test and the results are shown in
the paper. We used Phil Karn's software version of the RS decoder which has
the neat feature of reporting the number of bytes corrected in each block.
Plots of this parameter vs. time are interesting indicators of what is
happening to the link.

3. For multipath, a single carrier system needs an equalizer. Measurements
of the Delano to Washington path indicated that there were significant
multipath components most of the time. The delay spread was on the order of
1 millisecond. It would have been difficult to receive any reasonable
symbol rate over this channel without equalization.

The equalizer is computationally intensive and is one of the factors for
making the receiver work slowly. It would be nice to implement it into an
ASIC or DSP. The rest of the receiver is pretty straight forward.

Did someone mention a business proposition about making some money in the
amateur market? The major world short wave broadcasters are interested in
developing a digital system as well. There may be a market for receivers
there as well.

Arv

23. Re: [HFSIG:2595] DSPBOX program for noise reduction

From: Eduardo Jacob <edu@kender.es>
Date: Tue, 06 Jan 1998 22:01:19 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19980106220119.0082b6c0@kender.es>
Rendered body type: text/plain

>Thanks to Eduardo, EA2BAJ, the DSPBOX.ZIP file has been uploaded
>to the 'upload' directory of the TAPR ftp site. Now it should be moved
>to the right directory by the site manager (Greg may be ?)
>It is a program meant for the DSK 'C50, and I seem to remember that there
>should be a directory for this DSK somewhere.

Meanwhile it's available also at

ftp://det.bi.ehu.es/pub/ham/ti/Prog/Dspbox.zip

Btw, it works flawlessly.

Eduardo/EA2BAJ

24. Re: [HFSIG:2597] SW digital broadcast: (was Best Noise Reduction Algorithm)

From: Antony Martin <anthony.martin@gecm.com>
Date: Tue, 06 Jan 1998 23:00:35 +0000
To: hfsig@tapr.org
Message-ID: <980106230035.10064@oas01.gmcl.co.uk>
Rendered body type: text/plain

Thanks for your very informative posting Arvydas Vaisnys <vaisnys@zorba.jpl.nasa.gov>

...extensive content not repeated; refer previous post...

Can you respond to a couple of questions? I'm not a broadcast engineer
so I'm not in possession of all the facts:

The AM broadcast channel is currently 10kHz. As I understand there is
some plan, ruling or whatever by the international spectrum gods that all
SW broadcasting should move to SSB imminently to conserve bandwidth
and reduce spectrum pollution by reducing BW and necessary transmitter
powers.

How can an 8kbaud digital system be proposed if this is the case?

Can such a system compete with likely global satellite broadcasting,
eg. via the WORLDSPACE satellites in the near future? Digital audio
Satellite receivers using microwave frequencies will be much more
easily integrated and cost-reduced than shortwave receivers, and
will be unlikely to need dish antennas.
Satellite has advantages like continuous availability and more
bandwidth; whereas HF provides coverage to the listener for only part
of a day &/or needs frequency schedules.

Is HF likely to be able to compete with satellite in broadcast in the
next milennia?

Major listening audiences of SW broadcasting are in the far east and
3rd world; who's likely to fund the production of hi-tech receivers and
delivery of services? The US government? Does it propose to augment
or upgrade the existing VOA HF service for digital?

On a similar note, the UK's BBC has sold off the HF 'World Service'
network to raise cash for their plans to go digital, mainly the roll-out
of DAB (Digital Audio Broadcast) (VHF) and forthcoming DTT (Digital
Terrestrial Television) in the UK.

The standards for DAB and DTT are all set; DAB is available now if you
can afford the receivers; Major TV makers & chip manufacturers are
collaborating & investing in the necessary components for DTT.

Maybe there are amateurs that might like to try building a DAB receiver?
Personally I'd not consider it until it could be done with inexpensive
DSPs, like another 3 years time maybe, by which time dedicated chips
will flow in abundance and receivers will be so cheap in the shops.

Anthony.

A.N.Martin   E-mail anthony.martin@gecm.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK

25. Re: [HFSIG:2594] Re: Best Noise Reduction Algorithm

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 7 Jan 1998 15:57:55 -0800 (PST)
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <199801072357.PAA08506@servo.qualcomm.com>
Rendered body type: text/plain

>Have you heard the new stanag 4479 800b/s voice coder? It's really
>awful, almost entirely unintelligible IMO. If you thought LPC-10 2400b/s
>wasn't exactly broadcast quality, it's much worse. It's said to

I haven't heard STANAG 4479, but I have heard 2400 bps MELP. There's
a demo package for W95 on the net. See

http://www.aspi.com/speech/melpdemo.htm

While it definitely sounds like a vocoder, it is *much* better than
2400bps LPC-10. It's supposedly the new US military standard.

>Give me a higher coding rate, less FEC and a few more channel errors anyday.
>Uncoded 2400b/s speech does get better on good channels, but 800b/s speech
>with FEC sounds bad on all channels!!!

A fairer comparison would be SSB on a channel that's so bad that the
SSB user has to slow down and/or use phonetics to be understood. So if
SSB need not be real time, why not digital voice? Just buffer a
digital voice transmission made at a high enough data rate to be
intelligible (e.g., 2400bps with MELP) and send it at whatever data
rate the channel will support with the optimum amount of FEC. Then
play it back in real time at the receiver.

>Large interleave error correction that deals with HF fading can't be
>used on speech because of the delay is unacceptable. So there isn't much

>that FEC can do for speech on HF.

I disagree. Given that we're already talking half duplex, what's wrong
with a little extra delay if it dramatically improves the error rate?

>I'm thinking store-and-forward coded speech voicemail may be a good way
>to support speech on poor links? RAM is cheap these days...

Exactly.

Phil

26. Re: [HFSIG:2591] Re: Best Noise Reduction Algorithm

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 7 Jan 1998 16:18:21 -0800 (PST)
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <199801080018.QAA08691@servo.qualcomm.com>
Rendered body type: text/plain

>But only above threshold. I think the ear plus brain and an analog voice
>SSB signal still beat any digital scheme for weak-signal recovery. If
>anyone believes otherwise and has an algorithm, contact me and I'll propose
>a business arrangement. If it can be done, there is plenty of money to be
>made in the amateur radio market.

I disagree. Both analog and digital communication systems have
thresholds, and well-designed implementations of both can adapt to
poor channels such that they both operate above threshold. Skilled
SSB operators will slow down, repeat and use phonetics to be
understood on a bad channel. Digital communication systems do the same
things, calling them "rate reduction", "ARQ" and "FEC"
respectively. But the digital systems can implement more optimum
schemes that come far closer to the information theoretic limits than
the brain, ear and mouth can.

Yes, digital systems tend to have much more pronounced thresholds than
analog systems, but that's a direct and unavoidable result of
operating so much closer to the theoretical limit.

This is especially true on deep fading channels. A fade rate near the
syllabic rate of speech is especially disturbing to SSB users, but a
properly designed FEC scheme can take it in stride. Sure, it might
still not be able to handle real time digital voice at the quality
level you'd like -- but neither can SSB. So let's keep the comparisons
honest here.

I also disagree that there is plenty of money to be made in the amateur
radio market...

Phil

27. Re: [HFSIG:2596] Question on status of digital voice

From: Phil Karn <karn@qualcomm.com>
Date: Wed, 7 Jan 1998 16:29:01 -0800 (PST)
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <199801080029.QAA08736@servo.qualcomm.com>
Rendered body type: text/plain

>The audio quality is not good. It is quite understandable, but also
>quite harsh sounding and not pleasant to listen to. I'd rather listen
>to an SSB signal with the hiss and pop.

Sounds like the audio interface to the repeater isn't as good as it
should be. It could be as simple as improper frequency equalization.
Digital schemes work best when they operate end-to-end, without
having to splice them into existing analog schemes or converting
from one digital scheme to another.

>Some questions: Is this representative of low rate voice encoding
>that might also be used on HF? Are there other systems that might
>be tolerable to use? What's the state of the art?
>Consider this a general inquiry from someone who has no experience
>in this area.

I haven't kept track of all the various internet phone products out
there, but they use a mix of voice coders. The 13 kb/s GSM 06.10 coder
seems fairly popular, but there are others that sound better and

others that sound worse. Also, uncompensated packet losses on the
Internet can make digital speech sound pretty bad.

If you'd like to hear what a bunch of the more common vocoders sound
like, see my web page on the subject:

http://people.qualcomm.com/karn/voicedemo/index.html

--Phil

28. Re: [HFSIG:2599] Re: SW digital broadcast: (was Best Noise Reduction

From: Arvydas Vaisnys <vaisnys@zorba.jpl.nasa.gov>
Date: Thu, 08 Jan 1998 10:40:45 -0800
To: hfsig@tapr.org
Message-ID: <3.0.32.19980108104044.006c3198@zorba.jpl.nasa.gov>
Rendered body type: text/plain

At 05:09 PM 1/6/98 -0600, you wrote:
>Thanks for your very informative posting Arvydas Vaisnys
<vaisnys@zorba.jpl.nasa.gov>
>
>..extensive content not repeated; refer previous post...
>
>Can you respond to a couple of questions? I'm not a broadcast engineer
>so I'm not in possession of all the facts:
>
>The AM broadcast channel is currently 10kHz. As I understand there is
>some plan, ruling or whatever by the international spectrum gods that all
>SW broadcasting should move to SSB imminently to conserve bandwidth
>and reduce spectrum pollution by reducing BW and necessary transmitter
>powers.

The international spectrum gods (ITU-R) go through a spectrum reallocation
and rule making process during WARCs which are held every few years.
Fortunately this process can be influenced. If you come up with a "useful"
radiocommunication service and can get enough countries to vote with you,
you will get some spectrum for this service. An example is the allocation
of frequencies in the L and S bands for satellite and complementary
terrestrial digital sound broadcasting at the 1992 WARC.

I believe the current HF plan is on course toward 5 kHz SSB at some future
date like 2015, because to date that has seemed the best solution to short
wave broadcasting problems. If digital systems can be shown to be superior
to SSB, then this course will probably change and spectrum will be made
available for digital.

>How can an 8kbaud digital system be proposed if this is the case?
>
There is nothing sacred about 5 kHz or 10 kHz channels. If agreement is
reached that 20 kHz or some other bandwidth is best for digital
broadcasting, the rules will eventually change accordingly (hopefully).
Inertia and coordination issues with analog systems will initially tend to
keep the bandwidth at 10 kHz or less.

>Can such a system compete with likely global satellite broadcasting,
>eg. via the WORLDSPACE satellites in the near future? Digital audio
>Satellite receivers using microwave frequencies will be much more
>easily integrated and cost-reduced than shortwave receivers, and
>will be unlikely to need dish antennas.

I don't see any fundamental differences between digital satellite and
digital short wave receivers.

>Satellite has advantages like continuous availability and more
>bandwidth; whereas HF provides coverage to the listener for only part
>of a day &/or needs frequency schedules.

There are two advantages for HF over satellite:

1.a. Satellites cost a lot of money
1.b. The existing HF transmitter infrastructure can easily be converted to
digital broadcasting

2.a. You need permission to transmit into other countries via satellite
2.b. You don't need permission to transmit into other countries via HF (I
don't really know the rules, gentlemen's agreements, etc.). You can also
change your coverage area by slewing your antenna.

The one other competing service that offers such flexibility is the Internet.

>Is HF likely to be able to compete with satellite in broadcast in the
>next milennia?

Is my name Madame Esmeralda?

>Major listening audiences of SW broadcasting are in the far east and
>3rd world; who's likely to fund the production of hi-tech receivers and
>delivery of services? The US government? Does it propose to augment
>or upgrade the existing VOA HF service for digital?
>
The VOA is funding a small R&D effort to look at digital HF. If the results
are favorable, they will at least lobby at the ITU for this service.
International HF broadcasting is not a commercial service, so the money
will be made mainly by receiver manufacturers. Therefore industry has to
get behind the push for this service.

There is an industry/broadcaster consortium called DRM being formed, whose
goal is to "guide" the development of a digital HF standard. You can find
out more about it at http://www.drm.org/

>On a similar note, the UK's BBC has sold off the HF 'World Service'
>network to raise cash for their plans to go digital, mainly the roll-out
>of DAB (Digital Audio Broadcast) (VHF) and forthcoming DTT (Digital
>Terrestrial Television) in the UK.

Presumably whoever bought the HF 'World Service' is interested in making
money with it?

>The standards for DAB and DTT are all set; DAB is available now if you
>can afford the receivers; Major TV makers & chip manufacturers are
>collaborating & investing in the necessary components for DTT.

DAB is a trademark for the Eureka 147 system. I believe receivers are to
become available in mid 1998. Don't know at what initial cost. So if you
mean the standard for Eureka 147 ia all set, then yes. If you mean the
standard for digital audio broadcasting, then no. You mentioned the
Worldspace satellite system. In the U.S. there will also be two domestic
satellite broadcasting services. There is also a push to develop some in
band on channel (IBOC) systems for the FM and AM bands.

>Maybe there are amateurs that might like to try building a DAB receiver?
>Personally I'd not consider it until it could be done with inexpensive
>DSPs, like another 3 years time maybe, by which time dedicated chips
>will flow in abundance and receivers will be so cheap in the shops.
>
If amateurs can put up satellites, they could develop receivers. Actually
from what I see on this discussion group, there already is a lot of work
going on in digital HF systems. Whatever technology we developed is
available. As a start you can look at our test report paper, which I
offered last time.

Arv

>Anthony.
>
>____
>
>A.N.Martin   E-mail anthony.martin@gecm.com
>Snr. Development Engineer   Tel.   01245 353221 x4995
>Military Products Engineering   Fax.   01245 275993
>GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
>____
>
>
>

29. RE: [HFSIG:2600] Re: Best Noise Reduction Algorithm

From: "Brunson, Gordon R" <84817@drintuit.dr.lucent.com>
Date: Thu, 08 Jan 1998 14:18:46 -0700 (MST)
To: hfsig@tapr.org
Message-ID: <162945535@drintuit>
Rendered body type: text/plain

Phil & the group:
I am a systems engineer and have been in the voicemail industry
for over 15 years. I have followed HFSIG for a couple of years
now. Great Stuff. Your comment about poor HF channel conditions
resulting in a voicemail-like service caught my eye.

I believe that there is some middle ground between real-time
interractive conversations, and store-and-forward voice mail. Let
me explain.

While I agree that instantaneous error-free communications
is what everyone wants, most of us still have to put up with inconveniences
as the technology evolves. For example, lots of folks are talking about
multimedia messaging -- you know, get text mail, voicemail, faxmail, etc
from a single in-box using whatever access instrument (phone, laptop, etc) is
most convenient to you at the time -- So I have a laptop dialed into
a network over a modem link and I see my mailbox which is full of mail.
Some of them are long voicemail messages (5 minutes each lets say).
One of them might be from the boss. What do you do?

If you want error free playback, on any speed
CPU, you might be tempted to transfer the entire file and then begin
playback. Often this is a waste of time if the message contents turns
out not to be of particular use at the time - like it was a sales call and
not the boss.   Some would say disconnect the laptop and call in again
over the voicemail interface where you have the ability to quickly
navigate between messages. You'll hear just the parts of the messages
that are important to you, and skip the others. A better solution however is
to use an intermittent streaming approach on playback over a slow link.
This means that although the speech is broken at times, you maintain
your ability to navigate through it or skip it, and you have not lost
any intelligibility, although it is a bit annoying to have breaks in the
playback every 8 or 16 seconds.

Real-time communications are similar in many ways because it is the
ability to interract, change the flow of the discussion, respond, reply,
interrupt, etc. that we find pleasing. And these things can be preserved
if you essentially limit the amount voice that the system buffers (Beep
when the buffer is nearly full), and enforce a policy of intermittent link
turn-around checking (Red-light or different beep when there is incoming
traffic that you should pause for) Green-light or bleep when it's your
turn to talk, etc.

This type of channel will not support the kind of interractions that need to take
place when two people are arguing, but for polite conversation where
folks want to take turns, and listen intermittently to see if they are
being interrupted before proceeding with a lengthy transmission,
this could make the voice channel on HF essentially error-free from
an intelligibility perspective.

Thanks for listening.
Gordon Brunson KC0VC.

> Phil Karn
> Time/Date 08:42 AM 01/08/98.
>
 > I disagree. Given that we're already talking half duplex, what's wrong
> with a little extra delay if it dramatically improves the error rate?
>
>>I'm thinking store-and-forward coded speech voicemail may be a good way
>>to support speech on poor links? RAM is cheap these days...
>
> Exactly.
>
> Phil
>
>

30. Re: [HFSIG:2604] Re: Best Noise Reduction Algorithm

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 8 Jan 1998 18:45:50 -0800 (PST)
To: hfsig@tapr.org
Cc: karn@qualcomm.com
Message-ID: <199801090245.SAA20770@servo.qualcomm.com>
Rendered body type: text/plain

Well, I had a more narrow model in mind close to the way we hams
already talk over HF radio: half duplex, with transmissions of
technically arbitrary length but limited by custom. Push button, begin
talking. Radio buffers digitized speech as necessary if the encoder
rate is faster than the rate currently supported by the channel.
After you let go of the button, the radio continues to empty the
buffer (if necessary). Then the other guy pushes his button.

Because the RF modems could be turning around fairly rapidly during
actual transmission to provide link level ARQ, it would be easy to add
a "break" flag to the reply packets to get the attention of the
talking operator. This would be one feature not found in current SSB
radios, unless you consider short VOX timeouts and fast AGC recovery
that lets you hear somebody break in during a pause.

Phil

31. Re: [HFSIG:2603] Re: SW digital broadcast

From: Antony Martin <anthony.martin@gecm.com>
Date: Fri, 09 Jan 1998 13:12:23 +0000
To: hfsig@tapr.org
Message-ID: <980109131223.19083@oas01.gmcl.co.uk>
Rendered body type: text/plain

Thanks for your reply Arvydas Vaisnys <vaisnys@zorba.jpl.nasa.gov>

> ..extensive content not repeated; refer previous post...

DAB (eureka 147) receivers are available from people like Rhode &
Schwarz but they're not seen in the high street.

at GEC-Marconi we have been associated with the BBC in
commercialising their DAB research and Eddystone radio
are supplying DAB transmitter equipment.

The COFDM modulation is done by 4 Sharc DSPs. Prior to that are
separate musicam audio encoders.

>From memory, the DAB channel has around 1000 tones per symbol
at a symbol rate about 1000 /sec. You wind up with about 2
megabits raw data rate. Then theres some FEC and demultiplexing.
It encodes 4 or 8 stereo audio channels.

There may be enough MFLOPS in a floating-point C6x to demodulate
COFDM, otherwise a big FPGA may be better. Musicam is MPEG2 audio
so that isn't so much of a problem. Currently not cheap.

DAB has the advantage that transmitters for the same channels with
overlapping coverage areas can share the same frequency without
interference. For densely populated places like europe, lots of
frequencies are wasted in the analogue systems as they can't
be used in adjacent areas. DAB also copes well with the rapidly
varying multipath experienced by a moving vehicle in a cityscape.

> Presumably whoever bought the HF 'World Service' is interested in making
> money with it?
I think they're just leasing it back to the BBC for it's remaining life.
The BBC can raise some cash without getting into debt. Creative accountancy
I suppose.

> If amateurs can put up satellites, they could develop receivers. Actually

I get the feeling that an international multiband broadcast receiver
is likely to need to decode 6 or more digital formats covering DC to
daylight. Most receivers will be single-standard, so the international
traveller will get a raw deal. At least Eureka 147 appears to be
Europe-wide so long-distance truckdrivers & daytrippers will be ok!

Without multistandard receivers, what will become of the amateur sport
of DX in the year 2010...?

Anthony.

32. HF Radio News Item

From: Antony Martin <anthony.martin@gecm.com>
Date: Fri, 09 Jan 1998 13:33:25 +0000
To: hfsig@tapr.org
Message-ID: <980109133325.19083@oas01.gmcl.co.uk>
Rendered body type: text/plain

The UK coastguard has ceased monitoring the MF/HF morse
maritime distress channels.

HF morse distress has been replaced by HF DSC distress
watchkeeping receivers. There are now many other digital

means of distress alerting including Inmarsat satcomms
and EPIRBs (float-free satellite beacons). The IMO
GMDSS rules now mandate the use of these methods.

==========

Alas the call of SOS has passed into the annals of history.
Another feature of analogue HF comms, instituted in the
time of Marconi himself, and the time of the Titanic disaster,
and so well-known to the general public, is no more.

Anthony.

33. Re: [HFSIG:2607] HF Radio News Item

From: jsgamble@mindspring.com
Date: Fri, 09 Jan 1998 10:34:22 -0500
To: hfsig@tapr.org, hfsig@tapr.org
Message-ID: <3.0.5.32.19980109103422.007d5930@mindspring.com>
Rendered body type: text/plain

At 07:41 AM 1/9/98 -0600, Antony Martin wrote:
>The UK coastguard has ceased monitoring the MF/HF morse
>maritime distress channels.
>
>HF morse distress has been replaced by HF DSC distress
>watchkeeping receivers. There are now many other digital
>means of distress alerting including Inmarsat satcomms
>and EPIRBs (float-free satellite beacons). The IMO
>GMDSS rules now mandate the use of these methods.
>
>==========
>
>Alas the call of SOS has passed into the annals of history.
>Another feature of analogue HF comms, instituted in the
>time of Marconi himself, and the time of the Titanic disaster,
>and so well-known to the general public, is no more.
>
>Anthony.
>

   Let S.O.S. R.I.P.:>)
DE Joel,KO4QC

34. Re: [HFSIG:2606] Re: SW digital broadcast

From: Arvydas Vaisnys <vaisnys@zorba.jpl.nasa.gov>
Date: Fri, 09 Jan 1998 09:53:56 -0800
To: hfsig@tapr.org
Message-ID: <3.0.32.19980109095356.006ca480@zorba.jpl.nasa.gov>
Rendered body type: text/plain

At 07:16 AM 1/9/98 -0600, you wrote:
>DAB has the advantage that transmitters for the same channels with
>overlapping coverage areas can share the same frequency without
>interference. For densely populated places like europe, lots of
>frequencies are wasted in the analogue systems as they can't
>be used in adjacent areas. DAB also copes well with the rapidly
>varying multipath experienced by a moving vehicle in a cityscape.
>
An equalizer can also be used to allow multiple transmitters (single
frequency network) operation. Ditto multipath. The signal delay restriction
between transmitters is opposite that of DAB. More is better. Satellite
broadcasters may have to use on-channel terrestrial repeaters to ensure
good coverage. OFDM, equalization, and spread spectrum are all possible
solutions.

>I get the feeling that an international multiband broadcast receiver
>is likely to need to decode 6 or more digital formats covering DC to
>daylight. Most receivers will be single-standard, so the international
>traveller will get a raw deal. At least Eureka 147 appears to be
>Europe-wide so long-distance truckdrivers & daytrippers will be ok!
>
>Without multistandard receivers, what will become of the amateur sport
>of DX in the year 2010...?
>
Perhaps some future generation DSP will be able to handle all formats. And
some future generation A to D will allow sampling the entire RF spectrum
directly.

Anyway for the status of digital radio around the world see

http://www.magi.com/~moted/dr/
For a prediction of the future, see
http://www.bgsu.edu/Departments/tcom/nab-radio.html

Arv

>Anthony.

35. Re: [HFSIG:2594] HFSIG digest 465

From: CDrentea <CDrentea@aol.com>
Date: Sun, 11 Jan 1998 20:30:04 EST
To: hfsig@tapr.org
Message-ID: <dc2f2071.34b9721d@aol.com>
Rendered body type: text/plain

I bought the Book just prior to your e-mail by coincidence. It is just
delightful!

Best, Cornell

36. Re: DSP56002EVM radio interface

From: "Louis Cobet" <louis_cobet@email.msn.com>
Date: Sat, 17 Jan 1998 20:40:43 -0800
To: <hfsig@tapr.org>
Message-ID: <0262e2947041218UPIMSSMTPUSR01@email.msn.com>
Rendered body type: text/plain

I remember some talk many months ago about a radio interface for the
DSP56002EVM. There was some talk of a kit. So I have waited.

I would appreciate it if someone who has an interface of their own could
send me a schematic and I will build my own. I am not much of a digital
designer but I like to build.

Please contact me by email at the address below.

Louis Cobet, K6MDH
louis_cobet@email.msn.com

37. Re: [HFSIG:2611] Re: DSP56002EVM radio interface

From: forrerj@peak.org (Johan Forrer)
Date: Sat, 17 Jan 1998 23:02:30 -0800
To: hfsig@tapr.org
Message-ID: <199801180703.XAA00778@peak.org>
Rendered body type: text/plain

Louis,

>I remember some talk many months ago about a radio interface for the
>DSP56002EVM. There was some talk of a kit. So I have waited.
>
>I would appreciate it if someone who has an interface of their own could
>send me a schematic and I will build my own. I am not much of a digital
>designer but I like to build.
>
>Please contact me by email at the address below.
>
>Louis Cobet, K6MDH
>louis_cobet@email.msn.com

I don't quite know what is holding up the proposed TAPR EVM project. It
really is a very nicely done effort and quite simple to put together also
works just fine too.

Have you looked on the TAPR web page for shematics? Select DSP, follow the
56002EVM link and that will lead you to some of the hardware projects for
the EVM including schematics. If that does not help, e-mail me directly and
we can see what we can do to help.

73

--Johan, KC7WW

38. TOR modes for the EVM

From: forrerj@peak.org (Johan Forrer)
Date: Sat, 17 Jan 1998 23:22:24 -0800
To: DSP@tapr.org, HFSIG@tapr.org
Message-ID: <199801180723.XAA02323@peak.org>

Rendered body type: text/plain

Hi all,

I have uploaded EVMPAC3.ZIP to the recent_uploads area for experimenting
with. Look for it at ftp.tapr.org/tapr/SIG/dsp/recent_uploads.

The new version does not need any special RS232 control lines like RTS or
DTR as all communications to and from the EVM is via the SCI port at 19200.
These programs do however require some sort of interface to pull in PTT. I
can recommend using these programs with the N1OWU, DL loader.

The package includes source code for the HF modems for those that need to
modify them for their special hardware arrangements. Seems like a lot of
folks out there are doing their own interfacing.

If you are interested in trying Pactor or Amtor on your EVM, give these a try.

Enjoy!

--Johan, KC7WW

(PS. Tim, could you move the files please. There is some remnants called
EVMPAC2.ZIP that was left after I aborted the upload this morning. You can
safely delete that one.... The net was just too busy. Thanks much!)

39. qpsk on hf

From: Danie Brynard <BRYD@kidd.Co.Za>
Date: Mon, 19 Jan 1998 10:08:46 +0200
To: hfsig@tapr.org
Message-ID: <s4c32639.032@kidd.Co.Za>
Rendered body type: text/plain

I would like to make email contact (first) with hams who have used
newqpsk.asm successfully. (by Pawel) We tried it over the weekend
but could not get sync. I guess we are doing something wrong. Any tips
? Also my version could be old. I am using 22Aug95 version with
bios.asm of 1995 and also NOSVIEW.

I am also interested to do tests with hams in the rest of the world with
all the other modes ie slow psk ets. Please email me directly to arrange a
sked.

Danie zs6awk

40. [HFSIG:2613] TOR modes for the EVM -Reply

From: Danie Brynard <BRYD@kidd.Co.Za>
Date: Mon, 19 Jan 1998 11:04:49 +0200
To: hfsig@tapr.org
Message-ID: <s4c3336e.079@kidd.Co.Za>
Rendered body type: text/plain

Johan is it for tx and rx ?   Danie zs6awk

>>> Johan Forrer <forrerj@peak.org> 18/January/1998 09:30am >>>
Hi all,

I have uploaded EVMPAC3.ZIP to the recent_uploads area for
experimenting
with. Look for it at ftp.tapr.org/tapr/SIG/dsp/recent_uploads.

The new version does not need any special RS232 control lines like RTS
or
DTR as all communications to and from the EVM is via the SCI port at
19200.
These programs do however require some sort of interface to pull in
PTT. I
can recommend using these programs with the N1OWU, DL loader.

The package includes source code for the HF modems for those that
need to
modify them for their special hardware arrangements. Seems like a lot
of
folks out there are doing their own interfacing.

If you are interested in trying Pactor or Amtor on your EVM, give these a
try.

Enjoy!

--Johan, KC7WW

(PS. Tim, could you move the files please. There is some remnants
called
EVMPAC2.ZIP that was left after I aborted the upload this morning. You
can
safely delete that one.... The net was just too busy. Thanks much!)

41. DSPBOX

From: Alberto di Bene <dibene@usa.net>
Date: Mon, 19 Jan 1998 14:33:46 +0100
To: hfsig@tapr.org, dsp@tapr.org
Message-ID: <34C35639.1A3B1B14@usa.net>
Rendered body type: text/plain

Just a reminder, you can download DSPBOX.ZIP from the following location
:

    ftp://ftp.tapr.org/tapr/SIG/dsp/recent_uploads/dspbox.zip

What is DSPBOX ?

 DSPBOX is an attempt to code the Texas Instruments DSK 'C50 in such a
way
 to make it behave like a commercial DSP noise reduction unit, like the
 NIR-12, the TimeWave 599+ and others. Not only the noise reduction
function
 has been implemented, but automatic notch, AGC and Band Pass have as
well.

Reports on how you do you like (or not like) it, are welcome.

73s

Alberto, I2PHD

42. Re: [HFSIG:2615] TOR modes for the EVM -Reply

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 19 Jan 1998 08:32:05 -0800
To: hfsig@tapr.org
Message-ID: <199801191633.IAA14935@peak.org>
Rendered body type: text/plain

Danie,

>Johan is it for tx and rx ?   Danie zs6awk

EVMPAC does both RX and TX at 100 baud ASCII and Huffman.

BTW, WB6RIJ and I have had excellent results with Pawel's NEWQPSK on the
1500 mile path between us on 40 meters. We could even turn the power down
all the way and run on next to no power, at least no indication on the Bird
wattmeter. I also have tested it on the HF channel simulator all the way
down to +5 dB where it starts refusing to link. You are trying with
resonsable signals? Both ends of the link on the same sibeband and within
say 5 Hz of the frequency?

73

--Johan

>
>>>> Johan Forrer <forrerj@peak.org> 18/January/1998 09:30am >>>
>Hi all,
>
>I have uploaded EVMPAC3.ZIP to the recent_uploads area for
>experimenting
>with. Look for it at ftp.tapr.org/tapr/SIG/dsp/recent_uploads.
>
>The new version does not need any special RS232 control lines like RTS
>or
>DTR as all communications to and from the EVM is via the SCI port at
>19200.
>These programs do however require some sort of interface to pull in
>PTT. I
>can recommend using these programs with the N1OWU, DL loader.
>

>The package includes source code for the HF modems for those that
>need to
>modify them for their special hardware arrangements. Seems like a lot
>of
>folks out there are doing their own interfacing.
>
>If you are interested in trying Pactor or Amtor on your EVM, give these a
>try.
>
>Enjoy!
>
>--Johan, KC7WW
>
>(PS. Tim, could you move the files please. There is some remnants
>called
>EVMPAC2.ZIP that was left after I aborted the upload this morning. You
>can
>safely delete that one.... The net was just too busy. Thanks much!)
>
>
>
>

43. Harris 39 tone modems

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Mon, 19 Jan 1998 13:24:42 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.96.980119132234.20916J-100000@arctic>
Rendered body type: text/plain

Several Monthas ago, I picked up 5 of these (brand new) out of the 100 or
so batch that were being offered to the Ham COmmunity at $100 each. Did
the other 95 get sold? Anyone have them on the air? Mine are gathering
dust kinda waiting for someone else to do something...

de WB4APR

44. Re: [HFSIG:2600] Re: Best Noise Reduction Algorithm

From: Robert McGwier <rwmcgwier@worldnet.att.net>
Date: Tue, 27 Jan 1998 07:03:38 -0500
To: hfsig@tapr.org
Message-ID: <34CDCD1A.A4ABF74B@worldnet.att.net>
Rendered body type: text/plain

Phil Karn wrote:

> >Have you heard the new stanag 4479 800b/s voice coder? It's really
> >awful, almost entirely unintelligible IMO. If you thought LPC-10 2400b/s
> >wasn't exactly broadcast quality, it's much worse. It's said to
>
> I haven't heard STANAG 4479, but I have heard 2400 bps MELP. There's
> a demo package for W95 on the net. See
>
> http://www.aspi.com/speech/melpdemo.htm
>
> While it definitely sounds like a vocoder, it is *much* better than
> 2400bps LPC-10. It's supposedly the new US military standard.
>
> >Give me a higher coding rate, less FEC and a few more channel errors anyday.
> >Uncoded 2400b/s speech does get better on good channels, but 800b/s speech
> >with FEC sounds bad on all channels!!!
>
> A fairer comparison would be SSB on a channel that's so bad that the
> SSB user has to slow down and/or use phonetics to be understood. So if
> SSB need not be real time, why not digital voice? Just buffer a
> digital voice transmission made at a high enough data rate to be
> intelligible (e.g., 2400bps with MELP) and send it at whatever data
> rate the channel will support with the optimum amount of FEC. Then
> play it back in real time at the receiver.
>
> >Large interleave error correction that deals with HF fading can't be
> >used on speech because of the delay is unacceptable. So there isn't much
> >that FEC can do for speech on HF.
>
> I disagree. Given that we're already talking half duplex, what's wrong
> with a little extra delay if it dramatically improves the error rate?
>
> >I'm thinking store-and-forward coded speech voicemail may be a good way
> >to support speech on poor links? RAM is cheap these days...

>
> Exactly.
>
> Phil

I disagree as well that large interleavers introduce unacceptable delays on
channels that are so bad that you cannot use them otherwise. An
academic exercise is to understand convolutional interleavers as explained
in the STANAG documents that describe their MIL-STD-188-110A
cousin.

Bob

45. PACTOR QUESTION

From: "R. & S. Keller" <kellerrs@dpnet.net>
Date: Thu, 29 Jan 1998 00:49:50 -0600
To: "NETSIG list   mailing"<netsig@tapr.org>, " tapr-tnc "<tapr-tnc@tapr.org>,
        " Keith Justice "<kf7tp@goodnet.com>,
        "HF SIG list   mailing"<hfsig@tapr.org>
Message-ID: <v03130303b0f5d6dae6e8@[208.134.134.40]>
Rendered body type: text/plain

I'm trying to help a hearing impaired ham become active on Pactor. He is
running an ICOM IC-740 and we are having difficulty finding out how to
select the proper filter options for the rig. If anyone in TAPR is
currently running with this rig, would you please
*** contact me via e-mail (R. & S. Keller" <kellerrs@dpnet.net>) ***

Also, any information or experiences regarding how to better operate Pactor
for the hearing impaired would be greatly appreciated.

TNX and 73,

Bob
K3PCS

46. RE: [HFSIG:2620] PACTOR QUESTION

From: "Borcher, David (NG-EX)" <DBorcher@NLvl.com>
Date: Thu, 29 Jan 1998 08:52:08 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <c=US%a=_%p=GI%l=NOGALESXCH-980129135208Z-36580@htxchng0.gic.gi.com>
Rendered body type: text/plain

Hi Bob

I am not really familiar with the IC-740 and therefore don't know what
options might be available. I have some general observations that may
be helpful in your determinations of options and filters...

I operate Pactor I with a Yaesu FT900 and an AEA/Timewave DSP-232.

The critical aspects of Pactor are (1) the need to tune the signal very
precisely
(Pactor 1 requires tuning within about +/- 50 Hz), and (2) making sure
that the
receiver AGC is off or operating at its' fastest time constant. A 500-Hz
CW filter
selection is optimum, IF the filter can be used in the SSB/FSK mode and
IF the
proper audio tones can be derived thru the filter at equal amplitude. I
found that
my FT-900 won't allow the CW filter to be used in SSB transmit, and it's
almost
impossible to manage using the CW filter on receive and the SSB filter
on transmit,
while tuning the band. Also, my IF Shift / PBT won't really let me move
the filter to
the point where I can get good audio response of the Pactor tones. So I
picked
the DSP-232 for its' good filtering and "digital PLL" demodulation of
the Pactor I
FSK.

If you are speaking of using Pactor II, you're in another ball game -
you must keep
the signals tuned within about +/- 20 Hz, and you almost need an
IF/audio path
optimized for the DPSK modulation.   As well as the $950 for the modem.

Is this info of any help at all?

Dave K7NG

>-----Original Message-----
>From: R. & S. Keller [SMTP:kellerrs@dpnet.net]
>Sent: Wednesday, January 28, 1998 11:54 PM
>To: hfsig@tapr.org
>Subject: [HFSIG:2620] PACTOR QUESTION
>
>I'm trying to help a hearing impaired ham become active on Pactor. He is
>running an ICOM IC-740 and we are having difficulty finding out how to
>select the proper filter options for the rig. If anyone in TAPR is
>currently running with this rig, would you please
>*** contact me via e-mail (R. & S. Keller" <kellerrs@dpnet.net>) ***
>
>Also, any information or experiences regarding how to better operate Pactor
>for the hearing impaired would be greatly appreciated.
>
>TNX and 73,
>
>Bob
>K3PCS
>

47. Re: [HFSIG:2620] PACTOR QUESTION

From: w2pat@juno.com (Marvin Bernstein)
Date: Thu, 29 Jan 1998 11:06:05 -0500
To: hfsig@tapr.org
Message-ID: <19980129.110608.8150.1.w2pat@juno.com>
Rendered body type: text/plain

Answers:
1)logo-white print-CNR SERVES ALL CANADA(in white border box at 15 degree
angle) on green maple leaf herald.
2) green of maple leaf herald.
3)1940's.

how does something like " height of the steam era" sound?

_____
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Or call Juno at (800) 654-JUNO [654-5866]

48. APRS810

From: Joe & Bonita Sanada <heyco@esslink.com>
Date: Thu, 29 Jan 1998 12:21:10 -0500
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <34D0BA86.57791CBA@esslink.com>
Rendered body type: text/plain

Hi I have the latest version APRS810 and I also have a new vanity call
my old call was KA1ZXN my new one now is W1JAS how do i change the call
on the aprs810... Please let me know thanks 73 Joe W1JAS.

49. FS1015

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 30 Jan 1998 10:26:12 +0000
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <34D1AAC4.50B6279F@dircon.co.uk>
Rendered body type: text/plain

Hello,

I require a copy of FS1015, any suggestions on obtaining one,
cheaply (please bear in mind my U.K location)?

I think Federal Standards maybe copied freely?

Thanks.

- Charles G4GUO

February 1998

Original source file: hfsig.9802
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9802

1. FLASH for the ADSP2106 EVM ?

From: Danie Brynard <BRYD@kidd.Co.Za>
Date: Tue, 03 Feb 1998 08:10:22 +0200
To: hfsig@tapr.org
Message-ID: <s4d6ea43.082@kidd.Co.Za>
Rendered body type: text/plain

Did anybody devise a FLASH storage unit for the 'ADSP-21061 SHARC
EVM EZ-KIT Lite' yet ?

Danie

2. Startup HF Packet Station

From: "Ron Short" <rshort@whidbey.com>
Date: Tue, 3 Feb 1998 09:08:28 -0800
To: <hfsig@tapr.org>
Message-ID: <199802031707.JAA22191@whidbey.whidbey.com>
Rendered body type: text/plain

Hello,

I live in an area that there are very few HF packet stations. It is
difficult to find someone to help with setting up a station.

I have a ICOM 751-A transciever, PK-900 TNC running PC-Pakrat. I have
connected the TNC to the radio via the plastic plug in the back of the
radio. I assume that this is for JUST FSK operation. I must use the RTTY
button for this to work. I have been tols that this is the way to operate.
Well, I was asked to go to 14.061 mhz LSB (I think) and try a PACTOR
contact. I was not able to connect nor hear the station. I talked with him
on SSB voice and he said that I needed to go AFSK through the mike plug. I
am a bit confused. Thanks for any help that can be given.

e-mail to rshort@whidbey.com

73

KK7GO
Ron

3. HF Channel simulator demo available

From: forrerj@peak.org (Johan Forrer)
Date: Tue, 03 Feb 1998 14:20:09 -0800
To: HFSIG@tapr.org, DSP@tapr.org
Message-ID: <199802032221.OAA10459@peak.org>
Rendered body type: text/plain

Hi,

I have uploaded a demo of my EZ-KIT SHARC-based HF channel simulator for
those experimenting with HF.

ftp.tapr.org/tapr/SIG/dsp/recent_uploads/hfsimdemo.zip

The demo contains three examples: two CCIR POOR conditions, and one
passthrough. These all contain different SNR levels of Gaussian noise.

Sorry - no source code is available, but hope you find the demo of some use.

Enjoy!

Johan, KC7WW

(Tim, could you be so kind to move these files -- thanks much -JF.)

4. Updated version of DSPBOX

From: Alberto Di Bene <75800340@it.ibm.com>
Date: Sun, 15 Feb 1998 21:37:47 +0000
To: <dsp@tapr.org>
Cc: <hfsig@tapr.org>
Message-ID: <51200200036279010000002L212*@MHS>
Rendered body type: text/plain

I have just uploaded to /tapr/SIG/dsp/upload an amended version of DSPBOX.
The previous one had problems with fast Pentium machines.
Could please the site administrator move it to the right place ? Thanks..

It is also possible to download it from my page :
   http://www.qsl.net/i2phd

In case you don't know, DSPBOX is a program for the Texas 'C50 DSK,
to make it behave as a noise-reducing, multiple-notch, band-pass-filtering
unit. Comments and impressions are welcome.

Alberto di Bene, I2PHD   i2phd@qsl.net

5. Re: CCW vs. BPSK

From: "Louis Cobet" <louis_cobet@email.msn.com>
Date: Mon, 16 Feb 1998 17:34:01 -0800
To: <hfsig@tapr.org>
Message-ID: <066264236011128UPIMSSMTPUSR02@email.msn.com>
Rendered body type: text/plain

I am interested in the relative merits of CCW as against BPSK in weak signal
work. It seems to me that Coherent CW tied to a master clock could have some
real advantages in EME work. But it seems no one is experimenting with it.

Louis Cobet, K6MDH
louis_cobet@email.msn.com

6. Frequency vs. Propagation

From: RLANIER@mailb.harris.com (RLANIER)
Date: Fri, 20 Feb 1998 12:48:17 -0500
To: hfsig@tapr.org
Message-ID: <0004028E.1702@mailb.harris.com>
Rendered body type: text/plain

Hi all,

     I realize I should probably be shot for not knowing the answer to this
     question, but ...

     Is it a true statement that the shorter the wavelength, the shorter
     the propagation distance? In other words, 10GHz at 100 watts will only
     propagate 40 miles while 10MHz at 100 watts will propagate 400 miles,
     under the same conditions?

     This question stems from a discussion I had with my brother, who is a
     recent ham convert. We were discussing the merits of band planning and
     he asked me why weren't high-power stations moved up into the higher
     frequency bands and low-power stations moved into the lower frequency
     bands? Then I thought of the above statement, but I wasn't absolutely
     sure whether it was true.

     Tony

7. RE: [HFSIG:2630] Frequency vs. Propagation

From: "Borcher, David (NG-EX)" <DBorcher@gi.com>
Date: Fri, 20 Feb 1998 10:22:50 -0800
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <CDFDEF962C55D1119FC3006097C510D0198117@nogalesxch.147.84.168.IN-ADDR.ARPA>
Rendered body type: text/plain

Is it a true statement that the shorter the wavelength, the
shorter
             the propagation distance? In other words, 10GHz at 100
watts will only
             propagate 40 miles while 10MHz at 100 watts will propagate
400 miles,
             under the same conditions?

Hi Tony,

There is a straightforward calculation for free-space path loss that
will show you
the tradeoff between frequency and range.

L (dB) = 36.6 +20log D (Miles) + 20log F (MHz)

You can derive range vs. frequency and required ERP vs. range vs.
frequency, etc.

Note also that higher frequencies have the possibility of higher gain antennas on both ends
of the path, while you can generally run higher transmitter powers at lower frequencies.
And you might expect receiver sensitivity to be slightly better at lower frequencies,
discounting atmospheric and sky noise.

Of course, if you put an ionosphere in there, all bets are off, using the above formula.

73,
Dave K7NG

8. Re: [HFSIG:2630] Frequency vs. Propagation

From: Robert Glassey <robglassey@geocities.com>
Date: Sat, 21 Feb 1998 13:46:27 -0800
To: hfsig@tapr.org
Message-ID: <34EF4B33.4055@geocities.com>
Rendered body type: text/plain

> Is it a true statement that the shorter the wavelength, the shorter
> the propagation distance? In other words, 10GHz at 100 watts will only
> propagate 40 miles while 10MHz at 100 watts will propagate 400 miles,
> under the same conditions?

I gather this is true not only when pathloss calculations are considered, but also in many practical instances, when other factors come into play.

Long wave broadcast in Europe is a clear example. It's amazing how far a long wave station will propagate by groundwave compared to a medium wave station. This gives long wave stations 24 hour long distance coverage without relying on the ionosphere. As I understand it this is why long wavelengths were favoured in early radio communications, eg broadcast and shipping. I think it was the 24 hour reliability factor that was more important than the signal stength, those transmitters could almost ionize the upper atmosphere with pure RF energy! :-) Well, maybe not. Lower frequencies gave better 24 hour range for whatever power (in megawatts :-) you decided to use.

At higher frequencies, when we get back down to groundwave again, a 2m signal will go much further than a 23cm signal, given the same antenna radiation patern and gain. As frequency goes up the size of the RX antenna goes down, reducing effective capture area of the antenna, hence a higher apparent increase in pathloss for isotropic referenced antennas. But if you increase capture area, giving the antenna gain, you get back the extra pathloss. A large enough perfect dish will increase in antenna gain as the frequency gets higher just as fast as the "pathloss" increases, the net change in signal strength is 0dB. Your TX can still be isotropic, but your receiver becomes highly directive. Then you can start getting signal stength benifits by making your transmitter directive, and get even more gain. Of course the same applies to any pair of directive antennas. The higher the freq the more gain you get (for antennas of the same physical size), over and above increased pathloss. I think this is why 23cm EME is becomming more popular.

If the freqeuncy get too high of course, then water vapour and other gasses in the atmosphere start absorbing.

Back down to HF and things get more complicated, with antenna gain, antenna height in wavelengths, ionosphereic propagation and absorbtion, and atmospheric and cosmic noise, which all tend to make high frequencies better up to the point where the ionosphere stops reflecting the signals. And the more ionization, then higher the maximum frequency and the better the SNR at that freqeuency.

So if you're looking at groundwave propagation with low gain antennas, lower frequencies work better, but once you start bouncing the signals around or start using high gain antennas, then higher frequencies work better.

That's how I understand it anyway. Someone did ask...

Cheers,

Rob, now ZL2AKM

9. Re: [HFSIG:2630] Frequency vs. Propagation

From: Robert Glassey <robglassey@geocities.com>
Date: Sat, 21 Feb 1998 14:07:03 -0800
To: hfsig@tapr.org
Message-ID: <34EF5007.EAA@geocities.com>
Rendered body type: text/plain

> Is it a true statement that the shorter the wavelength, the shorter
> the propagation distance? In other words, 10GHz at 100 watts will only
> propagate 40 miles while 10MHz at 100 watts will propagate 400 miles,
> under the same conditions?

Just to put this in perspective, consider a 3 element 10MHz beam at 50 feet, and how far would you expect to get on ground wave? Then consider a 3 element 10GHz beam measureing about half an inch square, half an inch off the ground... Not exactly needle bending stuff! Now imagine a 500m diameter dish, 1km up in the air on 10MHz!

Cheers,

Rob

March 1998

Original source file: hfsig.9803
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9803

1. Visual C and RS232

From: Danie Brynard <BRYD@kidd.Co.Za>
Date: Mon, 02 Mar 1998 08:11:21 +0200
To: hfsig@tapr.org
Message-ID: <s4fa6c88.033@kidd.Co.Za>
Rendered body type: text/plain

Does anybody know where I can get an example on how to use the
RS232 ports in Visual C++ on Win95. I want to write a very simple
terminal for modem use.

Danie zs6awk

---

2. Beta of PSK31 with QPSK for Texas C50 Available!!

From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
Date: Tue, 03 Mar 1998 20:26:41 +0100
To: psk31@bipt106.bi.ehu.es
Message-ID: <3.0.5.32.19980303202641.00864b60@bicc00.bi.ehu.es>
Rendered body type: text/plain

There is a beta for PSK31 with QPSK for the DSKC50.
It's available at:

ftp://det.bi.ehu.es/pub/ham/psk31/psk31c50_beta1.zip

Comments are welcome.

        Eduardo/EA2BAJ

PD: PSK31 Information always available at
http://bipt106.bi.ehu.es/psk31.html

---

3. BPSK activities ?

From: "Lee, Choy Fook" <Cflee@vancouver.glenayre.com>
Date: Tue, 3 Mar 1998 12:35:04 -0800
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <A06FB716F2F3D01192DE00805F310441918CED@vancouver_nt2.vancouver.glenayre.com>
Rendered body type: text/plain

Are there any activities on BPSK lately and if on what freq and time
please?
73
ve7lcf

---

4. [psk31] PSK31C50 beta version bug!

From: Eduardo Jacob <edu@kender.es>
Date: Wed, 04 Mar 1998 22:56:08 -0100
To: dsp@tapr.org, hfsig@tapr.org
Message-ID: <3.0.5.32.19980304225608.00808210@kender.es>
Rendered body type: text/plain

>>> From Andrew Senior
Hello everyone,

If you are using the beta release of PSK31 for the 'C50, please note there is
at least one bug in it! I discovered this not long after I sent the package to
Eduardo to go on the web-site (oops!) The bug concerns the auto-CQ feature
(F5). It should work fine if you want to call CQ when you are not already
transmitting, but if you have already turned the transmitter on (i.e. pressed
F2), then press F5, although the first CQ call will go out fine, if you
subsequently come to press F5 to call again, the transmission will start
without a run of idles to allow the DCD to turn on. I have found and fixed
this bug and the first proper release will include this fix.

In the meantime, I recommend you steer clear of using auto-CQ when the
transmitter is already on, though you probably wouldn't want to do this very
often anyway.

Eduardo has asked me to implement the RTS PTT control and I will endeavour to
have this in the first proper release too.

73, Andrew.

G0TJZ

5. Turbo codes

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 20 Mar 1998 07:12:50 +0000
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <351216F2.36764946@dircon.co.uk>
Rendered body type: text/plain

Hello All,

I have been shown a good set of articles on Turbo Codes.
It is a four part series but only the first two parts are available
on line at present, the rest will follow. The magazine has a lot
of other interesting articles in it as well. Shame the paper based
version it is only free to subscribers in the US.

Call me mentally challenged or something but this is the first time
I have found an article about this subject that I can understand!

I await the final two instalments!

The URL is.

http://www.pein.com/

The editions you need are Jan and Feb.

- Charles G4GUO

6. HFSIG

From: Charles Brain <chbrain@dircon.co.uk>
Date: Sat, 21 Mar 1998 08:02:27 +0000
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <35137413.4233816@dircon.co.uk>
Rendered body type: text/plain

Hello Johan,

Have you been getting any HFSIG messages recently?
I sent one yesterday and never saw it comeback via the reflector?
It was about an easy to understand article on Turbo-coding.

I have ordered a student edition of Matlab via Amazon.com. I should
get it next week as they have said that it has been dispatched. What
I am going to do with it I don't know as I believe you have to get a
load of add on toolboxes before it is much use for our line of buisness.

- CUL Charles

7. Re: [HFSIG:2639] HFSIG

From: Gary Chatters <gc@universe.digex.net>
Date: Sat, 21 Mar 1998 18:46:00 -0500
To: hfsig@tapr.org
Cc: chbrain@dircon.co.uk
Message-ID: <199803212346.SAA02350@universe.digex.net>
Rendered body type: text/plain

Charles writes:
>
>Have you been getting any HFSIG messages recently?
>I sent one yesterday and never saw it comeback via the reflector?
>It was about an easy to understand article on Turbo-coding.
>

Sometime last month I noticed I wasn't getting any mail from HFSIG.
I checked my status and found that I was not subscribed to the list.
It turns out not much of anybody was. There were only about 15 or so
subscribers. I resubscribed, loosing only about 4 messages (not much
traffic in a month) and notice that the number of subscribers
is creeping back up. Almost 30 last time I checked.

There wasn't any mention of this on the list after I got resubscribed.
Was there a known problem that caused a loss of the subscriber list?

Gary

8. Being a Better Educator

From: pub@ge.net
Date: Sun, 22 Mar 1998 04:45:10 -0600 (CST)
To: <smcdermo@.calpoly.edu>
Message-ID: <199803221045.EAA13549@tapr.org>
Rendered body type: text/plain

Being a Better Educator
Dear educator:

Today, more than ever, we need innovative new solutions to
educational problems.

> "Kevin Kirchman's book Aspirations is an answer
> to education questions such as 'How do we teach
> students to think for themselves, fulfill their
> potential, be creative, gain self esteem, and be
> ready for this productive world?'
>
> "His book is a must for all educators from pre-school
> through graduate school. I wish his book had been
> available during my 42 year teaching career."
> Norma Silver
> Retired teacher, Fort Lee, NJ

Introducing,
 Aspirations: The Rational Foundations of Achievement.

The reason it is possible to have an entirely new perspective on
educational issues is because the theory behind education, called
epistemology, or the theory of knowledge, has _not_ been an
empirical
science.

Epistemology is to education what physics is to engineering--
but until the basis was discovered for _principles_ of
understanding,
there could be no science, and no applications that would
radically
transform educational practices.

That is until now.

> "If knowledge of the humanities is subjective, that is,
> only
> valid for the person who holds it, of what value is
> education?
>
> "Knowledge is not subjective. There are some conceptual
> models that are better--more accurate and enlightening--
> than others."
> Aspirations

Aspirations will show you why, and teach you precisely how to
tell the
difference.

> "[Aspirations] lays out the foundations of clear,
> critical
> thinking, that can help us to better understand
> ourselves,
> others, and the world around us. In the final analysis,
> we are
> all decision makers. Don't make many more important
> decisions
> before you read and grasp Aspirations."
> Frank L. David
> Co-Principal, Business Learning Centers
> Murrieta, CA

Perhaps the last earthly frontier, the mind, has been finally
penetrated by this wonderful new book. Aspirations won't give
you old ideas about the mind that you've seen before. It's
radically new.

Aspirations shows how, because of an old philosophical problem,
educators are unwittingly discouraging idea forming habits that
occur through the mental process of induction, or generalization.
The
book shows how these inductive concepts are the basis of all
abilities

and character.

Deterring concept formation has the effects of hindering individuality, stifling general competency and causing people to be more dependent upon the ideas formed by others (being less able to form them themselves). The book presents a new solution to this dilemma which describes the mental events that occur during concept formation that can be practiced and made habit.

Aspirations makes it clear that if you are less able to form concepts of your own, you necessarily have to borrow those formed by others. This discourages individuality, creativity, and self-reliance. It is also the basis for a host of psychological problems, from prejudice and intolerance to low self-esteem.

Set within the context of human intellectual history, Aspirations stands out as a unique and controversial contribution to our understanding of ourselves.

> Fascinating reading. A journey deep into the corners of the mind and down the paths of civilization's philosophical development.

The author of Aspirations, Kevin Kirchman, a Cornell University educated Artificial Intelligence scientist, lecturer and businessman, has actually utilized his new understanding of innovation to develop original perspectives on

* deduction, or what reasoning is
* induction, the basis of all innovation and creativity
* character formation

and these new ideas genuinely will aid you not only in teaching better mental habits, but in improving your own.

> "[Aspirations] is a self-motivational book for intelligent people."
>
> Stan Irwin
> The Producer of the Tonight Show
> with Johnny Carson for 15 years

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Aspirations will soon be promoted nationally by Rogers & Cowan, America's largest public relations firm. Be the first to learn and apply these revolutionary new insights.

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3 Concepts -- The Key to Understanding the Mind
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--
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Name ____________________________________________
Title ____________________
Organization_____________________________________
Mailing Addr _____________________________________
_____________________________________________
City ____________________ State _______________
Post Code _____________
Country _________________________

VISA ___   MasterCard ___   American Express ___ Discovery ___

Card #  __ __ __ __   __ __ __ __
   __ __ __ __   __ __ __ __

Expiration Date __________

Signature _____________________________________
```

9. Test-delete, please

```
From: PropNET <PropNET@greeceny.com>
Date: Fri, 27 Mar 1998 10:14:36 -0500
To: _APRS Listserve <aprssig@tapr.org>,
        _TAPR Freq Coord Listserve <regional_freq@tapr.org>,
        _TAPR HF Listserve <hfsig@tapr.org>
Message-ID: <351BC25C.61071ACB@greeceny.com>
Rendered body type: text/plain
```

Just testing 'cause I changed e-mail addresses, un-subscribed on the first, and re-subscribed on the second...with no traffic so far.
geez...some people can be so impatient :o)

10. ? Spec SlowBPSK ?

```
From: "Prof. Dr.-Ing. Wolf-Henning Rech" <rech@fh-pforzheim.de>
Date: Fri, 27 Mar 1998 16:30:52 +-100
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
```

Message-ID: <01BD599D.B74DAAE0@df9ic.ampr.org>
Rendered body type: text/plain

Hello,

I am looking for a spec of the slowbpsk transmission scheme.

I am not looking for DSP programs, source code, or a description what BPSK is and how it is implemented.

But I am interested if slowbpsk has a protocol, how it works, what type of coding (FEC ?) is used, what timing constraints exist - stuff like this.

I tried to find such information in the packet BBS and in the internet, but wasn´t yet successful.

Thanks for your help.

73 Henning

April 1998

Original source file: hfsig.9804
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9804

1. Release of Version 1.0 of PSK31 for the Texas C50 DSK.

From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
Date: Fri, 03 Apr 1998 12:45:50 +0200
To: psk31@bipt106.bi.ehu.es
Message-ID: <3.0.5.32.19980403124550.0080b690@bicc00.bi.ehu.es>
Rendered body type: text/plain

There is a new release of the Texas C50 PSK31 Terminal. It supports Viterbi
coding and QPSK modulation. These are the changes:

Changes since the Beta release:
* Serial port interrupt bug fixed: this caused the program to stop
  communicating with the DSK. (Didn't happen under Win3.x/Win95).
* Control of PTT via RTS line added.
* F5 key for auto-CQ now shown on display. Auto-CQ bug fixed.
* Faster tuning control (10 Hz steps with Ctrl+arrow keys)
* Tuning display direction of rotation changed (TAKE NOTE!)
* File send (F6) legend no longer shows 'on' or 'off'.
* Documentation updated.

As usual it's available at ftp://det.bi.ehu.es/pub/ham/psk31/psk31c50_r1.zip

If you want to know more about PSK31, please take a look at
http://aintel.bi.ehu.es/psk31.html

        Eduardo/EA2BAJ

Eduardo Jacob - Area de Ingenier'ia Telem'atica
Departamento de Electr'onica y Telecomunicaciones
ETSII y de IT   Tel: +34-(9)4-427 8055
UPV / EHU   Fax: +34-(9)4-441 4041
Alda Urquijo s/n   E-mail: jtpjatae@bi.ehu.es
E-48013 - Bilbao (Spain)   : edu@kender.es
Ham: EA2BAJ   VHF PACKET: EA2BAJ @ EA2URV
PGP Key available: Send email w/subject: PGPKEY
fingerprint   : D8 FA 54 49 F7 40 BA AA 09 1A 73 40 A4 26 ED BD

2. hfsig inactive?

From: Glenn Wiebe <gswiebe@mb.sympatico.ca>
Date: Mon, 13 Apr 1998 10:02:52 -0700
To: hfsig@tapr.org
Message-ID: <3532453C.50FF7E87@mb.sympatico.ca>
Rendered body type: text/plain

Hi!

I subscribed to the SIG on Friday but have received no mail yet. Does
this indicate that the SIG is dormant? It's a far cry from the stuff
that arrives daily from APRSSIG. :-)

73.....Glenn

Glenn Wiebe
Snail mail: Box 1563   Beausejour   MB   Canada   R0E 0C0
Email: gswiebe@mb.sympatico.ca
Ham radio call VE4GN   Packet: VE4GN@VE4UA.#WPG.MB.CAN.NOAM

3. Re: [HFSIG:2645] hfsig inactive?

From: Gary Chatters <gc@universe.digex.net>
Date: Mon, 13 Apr 1998 15:15:05 -0400
To: hfsig@tapr.org
Cc: gswiebe@mb.sympatico.ca
Message-ID: <199804131915.PAA03006@universe.digex.net>
Rendered body type: text/plain

>
>I subscribed to the SIG on Friday but have received no mail yet. Does
>this indicate that the SIG is dormant? It's a far cry from the stuff
>that arrives daily from APRSSIG. :-)
>

Traffic on the HFSIG list tends to be variable and rather low on the

average. Days may go by with no messages. A recent complicating
factor is that the subscription file got lost a couple of months ago.
After that the normally low traffic rate dropped even lower.
Last time I checked the subscriber list was up to about 50 so
we could be getting some more activity. Last previous message that
I received was on March 27, over two weeks ago. But the sequence number
indicates that I missed one.

Gary

4. Re: [HFSIG:2645] hfsig inactive?

From: Glenn Wiebe <gswiebe@mb.sympatico.ca>
Date: Mon, 13 Apr 1998 19:39:47 -0700
To: hfsig@tapr.org
Message-ID: <3532CC73.F8365E7B@mb.sympatico.ca>
Rendered body type: text/plain

Charles Brain wrote:

>
> Hello Glenn,
>
> No things have just been very quiet of late. I suggest you have
> a look through the archives to see what has been discussed in the
> past.
>
> - Charles G4GUO

Thanks for the tip on the archives, Charles. I checked it out and there
is certainly a lot of stuff there - most of it for rocket scientists.
Sigh! Oh well, I'll stick around for a bit and see what comes down the
pipe here in the sub-arctic. :-)

Gary - thanks for you comment.

73 all.....Glenn

--
Glenn Wiebe
Snail mail: Box 1563   Beausejour   MB   Canada   R0E 0C0
Email: gswiebe@mb.sympatico.ca
Ham radio call VE4GN   Packet: VE4GN@VE4UA.#WPG.MB.CAN.NOAM

5. Pactor x Clover ?

From: "Marcelo Puhl" <mark@plug-in.com.br>
Date: Mon, 13 Apr 1998 22:12:02 -3
To: hfsig@tapr.org
Message-ID: <199804140339.AAA25323@bigbox.plug-in.com.br>
Rendered body type: text/plain

Hello,

I would like to try Pactor or Clover for BBS forwarding on HF.

Which do you recommend ?

Where can I buy the hardware to use this modes ?

And how much I need to spend ?

Thanks for any info !

Marcelo - PY3SS
mark@plug-in.com.br

6. the list

From: Claude Frantz <claude@bauv106.bauv.unibw-muenchen.de>
Date: Tue, 14 Apr 1998 08:44:14 +0200 (MET DST)
To: hfsig@tapr.org
Message-ID: <199804140644.IAA20016@bauv106.bauv.unibw-muenchen.de>
Rendered body type: text/plain

Is the list still alive ?

7. Turbo codes for Ralyeigh channels

From: Tom McDermott <mcdermot@aud.alcatel.com>
Date: Tue, 14 Apr 1998 16:36:55 -0500

To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BD67C3.8AA69AE0.mcdermot@aud.alcatel.com>
Rendered body type: text/plain

I spotted an interesting article today:

"Design and Analysis of Turbo Codes on Rayleigh Fading Channels",
by Eric K. Hall and Stephen G. Wilson,
IEEE Journal on Selected Areas in Communications (JSAC),
Volume 16, Number 2, February 1998, pp 160-174.

Lot of good charts, but all based on simulation. Shows potentially
very impressive performance for Turbo-coded BPSK on Rayleigh channel.

Tom McDermott
mcdermot@aud.alcatel.com

8. [HFSIG:2637] [psk31] PSK31C50 beta version bug!

From: edu@kf9ug.#nin.in.usa.noam
Date: 04 Mar 98 22:07:00 GMT
To: hfsig@tapr.org
Message-ID: <26161edu@fwa.bbs>
Rendered body type: text/plain

R:980304/2207z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:15109 $:26161edu Z:46835

>>> From Andrew Senior
Hello everyone,

If you are using the beta release of PSK31 for the 'C50, please note there is
at least one bug in it! I discovered this not long after I sent the package to
Eduardo to go on the web-site (oops!) The bug concerns the auto-CQ feature
(F5). It should work fine if you want to call CQ when you are not already
transmitting, but if you have already turned the transmitter on (i.e. pressed
F2), then press F5, although the first CQ call will go out fine, if you
subsequently come to press F5 to call again, the transmission will start
without a run of idles to allow the DCD to turn on. I have found and fixed
this bug and the first proper release will include this fix.

In the meantime, I recommend you steer clear of using auto-CQ when the
transmitter is already on, though you probably wouldn't want to do this very
often anyway.

Eduardo has asked me to implement the RTS PTT control and I will endeavour to
have this in the first proper release too.

73, Andrew.
G0TJZ

9. [HFSIG:2634] Visual C and RS232

From: bryd@kf9ug.#nin.in.usa.noam
Date: 02 Mar 98 06:22:00 GMT
To: hfsig@tapr.org
Message-ID: <23448bryd@fwa.bbs>
Rendered body type: text/plain

R:980302/0622z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:7760 $:23448BRYD Z:46835

Does anybody know where I can get an example on how to use the
RS232 ports in Visual C++ on Win95. I want to write a very simple
terminal for modem use.

Danie zs6awk

10. [HFSIG:2635] Beta of PSK31 with QPSK for Texas C50 Available!!

From: jtpjat@kf9ug.#nin.in.usa.noam
Date: 03 Mar 98 19:39:00 GMT
To: hfsig@tapr.org
Message-ID: <51781jtpjat@fwa.bbs>
Rendered body type: text/plain

R:980303/1939z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:12010 $:51781jtpjat Z:46835

There is a beta for PSK31 with QPSK for the DSKC50.
It's available at:

ftp://det.bi.ehu.es/pub/ham/psk31/psk31c50_beta1.zip

Comments are welcome.

Eduardo/EA2BAJ

PD: PSK31 Information always available at
http://bipt106.bi.ehu.es/psk31.html

11. [HFSIG:2638] Turbo codes

From: chbrai@kf9ug.#nin.in.usa.noam
Date: 20 Mar 98 07:26:00 GMT
To: hfsig@tapr.org
Message-ID: <41858chbrai@fwa.bbs>
Rendered body type: text/plain

R:980320/0726z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:28605 $:41858chbrai Z:46835

Hello All,

I have been shown a good set of articles on Turbo Codes.
It is a four part series but only the first two parts are available
on line at present, the rest will follow. The magazine has a lot
of other interesting articles in it as well. Shame the paper based
version it is only free to subscribers in the US.

Call me mentally challenged or something but this is the first time
I have found an article about this subject that I can understand!

I await the final two instalments!

The URL is.

http://www.pein.com/

The editions you need are Jan and Feb.

- Charles G4GUO

12. [HFSIG:2640] Re: HFSIG

From: gc@kf9ug.#nin.in.usa.noam
Date: 21 Mar 98 23:59:00 GMT
To: hfsig@tapr.org
Message-ID: <22870gc@fwa.bbs>
Rendered body type: text/plain

R:980321/2359z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:29887 $:22870gc Z:46835

Charles writes:
>
>Have you been getting any HFSIG messages recently?
>I sent one yesterday and never saw it comeback via the reflector?
>It was about an easy to understand article on Turbo-coding.
>

Sometime last month I noticed I wasn't getting any mail from HFSIG.
I checked my status and found that I was not subscribed to the list.
It turns out not much of anybody was. There were only about 15 or so
subscribers. I resubscribed, loosing only about 4 messages (not much
traffic in a month) and notice that the number of subscribers
is creeping back up. Almost 30 last time I checked.

There wasn't any mention of this on the list after I got resubscribed.
Was there a known problem that caused a loss of the subscriber list?

Gary

13. [HFSIG:2636] BPSK activities ?

From: cflee@kf9ug.#nin.in.usa.noam
Date: 03 Mar 98 20:48:00 GMT
To: hfsig@tapr.org
Message-ID: <3849cflee@fwa.bbs>
Rendered body type: text/plain

R:980303/2048z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:12148 $:3849Cflee Z:46835

Are there any activities on BPSK lately and if on what freq and time
please?
73
ve7lcf

14. [HFSIG:2639] HFSIG

From: chbrai@kf9ug.#nin.in.usa.noam
Date: 21 Mar 98 08:17:00 GMT
To: hfsig@tapr.org
Message-ID: <37237chbrai@fwa.bbs>
Rendered body type: text/plain

R:980321/0817z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:29577 $:37237chbrai Z:46835

Hello Johan,

Have you been getting any HFSIG messages recently?
I sent one yesterday and never saw it comeback via the reflector?
It was about an easy to understand article on Turbo-coding.

I have ordered a student edition of Matlab via Amazon.com. I should
get it next week as they have said that it has been dispatched. What
I am going to do with it I don't know as I believe you have to get a
load of add on toolboxes before it is much use for our line of buisness.

- CUL Charles

15. [HFSIG:2643] ? Spec SlowBPSK ?

From: rech@kf9ug.#nin.in.usa.noam
Date: 27 Mar 98 19:14:00 GMT
To: hfsig@tapr.org
Message-ID: <37523rech@fwa.bbs>
Rendered body type: text/plain

R:980327/1914z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:34242 $:37523rech Z:46835

Hello,

I am looking for a spec of the slowbpsk transmission scheme.

I am not looking for DSP programs, source code, or a description what =
BPSK
is and how it is implemented.

But I am interested if slowbpsk has a protocol, how it works, what type =
of
coding (FEC ?) is used, what timing constraints exist - stuff like this.

I tried to find such information in the packet BBS and in the internet, =
but
wasn=B4t yet successful.

Thanks for your help.

73   Henning

16. [HFSIG:2647] Re: hfsig inactive?

From: gswieb@kf9ug.#nin.in.usa.noam
Date: 14 Apr 98 01:00:00 GMT
To: hfsig@tapr.org
Message-ID: <44097gswieb@fwa.bbs>
Rendered body type: text/plain

R:980414/0100z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:44031 $:44097gswieb Z:46835

Charles Brain wrote:

>
> Hello Glenn,
>
> No things have just been very quiet of late. I suggest you have
> a look through the archives to see what has been discussed in the
> past.
>
> - Charles G4GUO

Thanks for the tip on the archives, Charles. I checked it out and there
is certainly a lot of stuff there - most of it for rocket scientists.
Sigh! Oh well, I'll stick around for a bit and see what comes down the
pipe here in the sub-arctic. :-)

Gary - thanks for you comment.

73 all.....Glenn

```
--
Glenn Wiebe
Snail mail: Box 1563   Beausejour   MB   Canada   R0E 0C0
Email: gswiebe@mb.sympatico.ca
Ham radio call VE4GN   Packet: VE4GN@VE4UA.#WPG.MB.CAN.NOAM
```

17. [HFSIG:2641] Being a Better Educator

```
From: pub@kf9ug.#nin.in.usa.noam
Date: 22 Mar 98 10:59:00 GMT
To: hfsig@tapr.org
Message-ID: <55908pub@fwa.bbs>
Rendered body type: text/plain
```

R:980322/1059z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:30167 $:55908pub Z:46835

```
Being a Better Educator
Dear educator:

Today, more than ever, we need innovative new solutions to
educational problems.

        "Kevin Kirchman's book Aspirations is an answer
        to education questions such as 'How do we teach
        students to think for themselves, fulfill their
        potential, be creative, gain self esteem, and be
        ready for this productive world?'

        "His book is a must for all educators from pre-school
        through graduate school. I wish his book had been
        available during my 42 year teaching career."
                 Norma Silver
                 Retired teacher, Fort Lee, NJ

Introducing,
         Aspirations: The Rational Foundations of Achievement.

The reason it is possible to have an entirely new perspective on
educational issues is because the theory behind education, called
epistemology, or the theory of knowledge, has _not_ been an
empirical
science.

Epistemology is to education what physics is to engineering--
but until the basis was discovered for _principles_ of
understanding,
there could be no science, and no applications that would
radically
transform educational practices.

That is until now.

        "If knowledge of the humanities is subjective, that is,
only
        valid  for the person who holds it, of what value is
        education?

        "Knowledge is not subjective. There are some conceptual
        models that are better--more accurate and enlightening--
        than others."
                Aspirations

Aspirations will show you why, and teach you precisely how to
tell the
difference.

        "[Aspirations] lays out the foundations of clear,
critical
        thinking, that can help us to better understand
ourselves,
        others, and the world around us. In the final analysis,
we are
        all decision makers. Don't make many more important
decisions
        before you read and grasp Aspirations."
                Frank L. David
                        Co-Principal, Business Learning Centers
                        Murrieta, CA
```

Perhaps the last earthly frontier, the mind, has been finally penetrated by this wonderful new book. Aspirations won't give you old ideas about the mind that you've seen before. It's radically new.

Aspirations shows how, because of an old philosophical problem, educators are unwittingly discouraging idea forming habits that occur through the mental process of induction, or generalization. The book shows how these inductive concepts are the basis of all abilities and character.

Deterring concept formation has the effects of hindering individuality, stifling general competency and causing people to be more dependent upon the ideas formed by others (being less able to form them themselves). The book presents a new solution to this dilemma which describes the mental events that occur during concept formation that can be practiced and made habit.

Aspirations makes it clear that if you are less able to form concepts of your own, you necessarily have to borrow those formed by others. This discourages individuality, creativity, and self-reliance.
It is also the basis for a host of psychological problems, from prejudice and intolerance to low self-esteem.

Set within the context of human intellectual history, Aspirations stands out as a unique and controversial contribution to our understanding of ourselves.

> Fascinating reading. A journey deep into the corners of the mind and down the paths of civilization□s philosophical development.

The author of Aspirations, Kevin Kirchman, a Cornell University educated Artificial Intelligence scientist, lecturer and businessman, has actually utilized his new understanding of innovation to develop original perspectives on

* deduction, or what reasoning is
* induction, the basis of all innovation and creativity
* character formation

and these new ideas genuinely will aid you not only in teaching better mental habits, but in improving your own.

> "[Aspirations] is a self-motivational book for intelligent people."
>
> Stan Irwin
> The Producer of the Tonight Show
> with Johnny Carson for 15 years

Rarely in human history is a book produced which has as important implications. Rarely is a book offered which gives us such hope and inspiration.

Aspirations will soon be promoted nationally by Rogers & Cowan, America□s largest public relations firm. Be the first to learn and apply these revolutionary new insights.

Table of Contents:

1 Introduction
2 The Problem -- Why Educational Philosophy is Psychologically Debilitating
3 Concepts -- The Key to Understanding the Mind
4 Descriptive Concepts -- The Foundation of Clear Thinking
5 Creating Ideas -- Concept Formation is Induction

```
         6    Character -- The Conceptual Origins of Personal
                  Qualities
         7    Motivation -- A Theoretical Defense of the
                  Principles of Achievement
         8    Reasoning -- A New and Practical Overview
         9    Goal Setting and Decision Making -- Giving
                  Yourself a Purpose
        10    Deduction -- The Connection Between Logic and
                  Common Sense
        11    Reasoning Well -- Applying the New Science of Logic
        12    The Principles of Induction -- New Solutions
                  to a Classical Challenge
        13    Conceptual Modeling -- Coming Up with the Best
                  Concepts
        14    Teaching Induction -- Overcoming the Fear of
                  Abstractions
        15    Philosophical Clarifications -- Understanding
                  the Breakthrough and Replacing Outdated
                  Technology
        16    The New Science of Logic -- Clearing the Way for a
                  Revolution in Rationality
        17    Historical Background and Philosophical Comparison
--
                  Putting the Conceptual Model Theory Into
                  Perspective

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Post Code _____________
Country _________________________

VISA ___   MasterCard ___   American Express ___ Discovery ___

Card #  __ __ __ __   __ __ __ __
   __ __ __ __   __ __ __ __

Expiration Date __________

Signature _____________________________________
```

18. [HFSIG:2648] Pactor x Clover ?

```
From: mark@kf9ug.#nin.in.usa.noam
Date: 14 Apr 98 03:58:00 GMT
To: hfsig@tapr.org
Message-ID: <59680mark@fwa.bbs>
Rendered body type: text/plain
```

R:980414/0358z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:44086 $:59680mark Z:46835

Hello,

I would like to try Pactor or Clover for BBS forwarding on HF.

Which do you recommend ?

Where can I buy the hardware to use this modes ?

And how much I need to spend ?

Thanks for any info !

Marcelo - PY3SS
mark@plug-in.com.br

19. [HFSIG:2650] Turbo codes for Ralyeigh channels

From: mcderm@kf9ug.#nin.in.usa.noam
Date: 14 Apr 98 21:57:00 GMT
To: hfsig@tapr.org
Message-ID: <21470mcderm@fwa.bbs>
Rendered body type: text/plain

R:980414/2157z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:44501 $:21470mcderm Z:46835

I spotted an interesting article today:

"Design and Analysis of Turbo Codes on Rayleigh Fading Channels",
by Eric K. Hall and Stephen G. Wilson,
IEEE Journal on Selected Areas in Communications (JSAC),
Volume 16, Number 2, February 1998, pp 160-174.

Lot of good charts, but all based on simulation. Shows potentially
very impressive performance for Turbo-coded BPSK on Rayleigh channel.

Tom McDermott
mcdermot@aud.alcatel.com

20. [HFSIG:2649] the list

From: claude@kf9ug.#nin.in.usa.noam
Date: 14 Apr 98 07:10:00 GMT
To: hfsig@tapr.org
Message-ID: <61306claude@fwa.bbs>
Rendered body type: text/plain

R:980414/0710z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:44151 $:61306claude Z:46835

Is the list still alive ?

21. [HFSIG:2646] Re: hfsig inactive?

From: gc@kf9ug.#nin.in.usa.noam
Date: 13 Apr 98 19:38:00 GMT
To: hfsig@tapr.org
Message-ID: <48882gc@fwa.bbs>
Rendered body type: text/plain

R:980413/1938z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:43881 $:48882gc Z:46835

>
>I subscribed to the SIG on Friday but have received no mail yet. Does
>this indicate that the SIG is dormant? It's a far cry from the stuff
>that arrives daily from APRSSIG. :-)
>

Traffic on the HFSIG list tends to be variable and rather low on the
average. Days may go by with no messages. A recent complicating
factor is that the subscription file got lost a couple of months ago.
After that the normally low traffic rate dropped even lower.
Last time I checked the subscriber list was up to about 50 so
we could be getting some more activity. Last previous message that
I received was on March 27, over two weeks ago. But the sequence number
indicates that I missed one.

Gary

22. [HFSIG:2642] Test-delete, please

From: propne@kf9ug.#nin.in.usa.noam
Date: 27 Mar 98 19:19:00 GMT
To: hfsig@tapr.org
Message-ID: <4981propne@fwa.bbs>
Rendered body type: text/plain

R:980327/1919z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:34250 $:4981PropNE Z:46835

Just testing 'cause I changed e-mail addresses, un-subscribed on the
first, and re-subscribed on the second...with no traffic so far.
geez...some people can be so impatient :o)

23. [HFSIG:2645] hfsig inactive?

From: gswieb@kf9ug.#nin.in.usa.noam
Date: 13 Apr 98 19:07:00 GMT
To: hfsig@tapr.org
Message-ID: <56693gswieb@fwa.bbs>
Rendered body type: text/plain

R:980413/1907z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:43857 $:56693gswieb Z:46835

Hi!

I subscribed to the SIG on Friday but have received no mail yet. Does
this indicate that the SIG is dormant? It's a far cry from the stuff
that arrives daily from APRSSIG. :-)

73.....Glenn

Glenn Wiebe
Snail mail: Box 1563   Beausejour   MB   Canada   R0E 0C0
Email: gswiebe@mb.sympatico.ca
Ham radio call VE4GN   Packet: VE4GN@VE4UA.#WPG.MB.CAN.NOAM

24. [HFSIG:2629] Re: CCW vs. BPSK

From: louis_@kf9ug.#nin.in.usa.noam
Date: 17 Feb 98 10:52:00 GMT
To: hfsig@tapr.org
Message-ID: <28575louis_@fwa.bbs>
Rendered body type: text/plain

R:980217/1052z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:84730 $:28575louis_ Z:46835

I am interested in the relative merits of CCW as against BPSK in weak signal
work. It seems to me that Coherent CW tied to a master clock could have some
real advantages in EME work. But it seems no one is experimenting with it.

Louis Cobet, K6MDH
louis_cobet@email.msn.com

25. [HFSIG:2628] Updated version of DSPBOX

From: 758003@kf9ug.#nin.in.usa.noam
Date: 15 Feb 98 21:45:00 GMT
To: hfsig@tapr.org
Message-ID: <7427758003@fwa.bbs>
Rendered body type: text/plain

R:980215/2145z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:83763 $:7427758003 Z:46835

I have just uploaded to /tapr/SIG/dsp/upload an amended version of DSPB=
OX.
The previous one had problems with fast Pentium machines.
Could please the site administrator move it to the right place ? Thanks=
..

It is also possible to download it from my page :
   http://www.qsl.net/i2phd

In case you don't know, DSPBOX is a program for the Texas 'C50 DSK,
to make it behave as a noise-reducing, multiple-notch, band-pass-filter=
ing
unit. Comments and impressions are welcome.

Alberto di Bene, I2PHD   i2phd@qsl.net
=

26. [HFSIG:2631] RE: Frequency vs. Propagation

From: dborch@kf9ug.#nin.in.usa.noam
Date: 20 Feb 98 19:54:00 GMT
To: hfsig@tapr.org
Message-ID: <59764dborch@fwa.bbs>
Rendered body type: text/plain

R:980220/1954z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:88373 $:59764DBorch Z:46835

Is it a true statement that the shorter the wavelength, the
shorter
the propagation distance? In other words, 10GHz at 100
watts will only
propagate 40 miles while 10MHz at 100 watts will propagate
400 miles,
under the same conditions?

Hi Tony,

There is a straightforward calculation for free-space path loss that
will show you
the tradeoff between frequency and range.

L (dB) = 36.6 +20log D (Miles) + 20log F (MHz)

You can derive range vs. frequency and required ERP vs. range vs.
frequency, etc.

Note also that higher frequencies have the possibility of higher gain
antennas on both ends
of the path, while you can generally run higher transmitter powers at
lower frequencies.
And you might expect receiver sensitivity to be slightly better at lower
frequencies,
discounting atmospheric and sky noise.

Of course, if you put an ionosphere in there, all bets are off, using
the above formula.

73,
Dave K7NG

27. [HFSIG:2630] Frequency vs. Propagation

From: rlanie@kf9ug.#nin.in.usa.noam
Date: 20 Feb 98 19:55:00 GMT
To: hfsig@tapr.org
Message-ID: <34261rlanie@fwa.bbs>
Rendered body type: text/plain

R:980220/1955z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:88379 $:34261RLANIE Z:46835

Hi all,

I realize I should probably be shot for not knowing the answer to this
question, but ...

Is it a true statement that the shorter the wavelength, the shorter
the propagation distance? In other words, 10GHz at 100 watts will only
propagate 40 miles while 10MHz at 100 watts will propagate 400 miles,
under the same conditions?

This question stems from a discussion I had with my brother, who is a
recent ham convert. We were discussing the merits of band planning and
he asked me why weren't high-power stations moved up into the higher
frequency bands and low-power stations moved into the lower frequency
bands? Then I thought of the above statement, but I wasn't absolutely
sure whether it was true.

Tony

28. [HFSIG:2633] Re: Frequency vs. Propagation

From: robgla@kf9ug.#nin.in.usa.noam
Date: 21 Feb 98 01:12:00 GMT
To: hfsig@tapr.org
Message-ID: <19258robgla@fwa.bbs>
Rendered body type: text/plain

R:980221/0112z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:89407 $:19258robgla Z:46835

>   Is it a true statement that the shorter the wavelength, the shorter

> the propagation distance? In other words, 10GHz at 100 watts will only
> propagate 40 miles while 10MHz at 100 watts will propagate 400 miles,
> under the same conditions?

Just to put this in perspective, consider a 3 element 10MHz beam at 50 feet, and how far would you expect to get on ground wave? Then consider a 3 element 10GHz beam measureing about half an inch square, half an inch off the ground... Not exactly needle bending stuff! Now imagine a 500m diameter dish, 1km up in the air on 10MHz!

Cheers,

Rob

29. [HFSIG:2632] Re: Frequency vs. Propagation

From: robgla@kf9ug.#nin.in.usa.noam
Date: 21 Feb 98 01:12:00 GMT
To: hfsig@tapr.org
Message-ID: <15592robgla@fwa.bbs>
Rendered body type: text/plain

R:980221/0112z @:KF9UG.#NIN.IN.USA.NOAM [EN71LD] #:89412 $:15592robgla Z:46835

> Is it a true statement that the shorter the wavelength, the shorter
> the propagation distance? In other words, 10GHz at 100 watts will only
> propagate 40 miles while 10MHz at 100 watts will propagate 400 miles,
> under the same conditions?

I gather this is true not only when pathloss calculations are considered, but also in many practical instances, when other factors come into play.

Long wave broadcast in Europe is a clear example. It's amazing how far a long wave station will propagate by groundwave compared to a medium wave station. This gives long wave stations 24 hour long distance coverage without relying on the ionosphere. As I understand it this is why long wavelengths were favoured in early radio communications, eg broadcast and shipping. I think it was the 24 hour reliability factor that was more important than the signal stength, those transmitters could almost ionize the upper atmosphere with pure RF energy! :-) Well, maybe not. Lower frequencies gave better 24 hour range for whatever power (in megawatts :-) you decided to use.

At higher frequencies, when we get back down to groundwave again, a 2m signal will go much further than a 23cm signal, given the same antenna radiation patern and gain. As frequency goes up the size of the RX antenna goes down, reducing effective capture area of the antenna, hence a higher apparent increase in pathloss for isotropic referenced antennas. But if you increase capture area, giving the antenna gain, you get back the extra pathloss. A large enough perfect dish will increase in antenna gain as the frequency gets higher just as fast as the "pathloss" increases, the net change in signal strength is 0dB. Your TX can still be isotropic, but your receiver becomes highly directive. Then you can start getting signal stength benifits by making your transmitter directive, and get even more gain. Of course the same applies to any pair of directive antennas. The higher the freq the more gain you get (for antennas of the same physical size), over and above increased pathloss. I think this is why 23cm EME is becomming more popular.

If the freqeuncy get too high of course, then water vapour and other gasses in the atmosphere start absorbing.

Back down to HF and things get more complicated, with antenna gain, antenna height in wavelengths, ionosphereic propagation and absorbtion, and atmospheric and cosmic noise, which all tend to make high frequencies better up to the point where the ionosphere stops reflecting the signals. And the more ionization, then higher the maximum frequency and the better the SNR at that freqeuency.

So if you're looking at groundwave propagation with low gain antennas, lower frequencies work better, but once you start bouncing the signals around or start using high gain antennas, then higher frequencies work better.

That's how I understand it anyway. Someone did ask...

Cheers,

Rob, now ZL2AKM

May 1998

Original source file: hfsig.9805
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9805

1. HF to Internet

From: The Whiting Household <rwhiting@winternet.com>
Date: Fri, 01 May 1998 19:18:41 -0500
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19980501191841.007d33a0@mail.winternet.com>
Rendered body type: text/plain

Please respond directly to Ben (see below).

Cheers/ Rick W0TN

From: Ben Parker [SMTP:ben@ocha.unon.org]
Sent: Tuesday, April 28, 1998 5:37 AM
To: gwyn@paccomm.com
Subject: [GATEWAY:182] HF radio e-mail

Apologies to those who have already heard from me.

I wonder if you could take the time to let me know what hardware and
software you are aware of that achieves e-mail (to and from Internet) over
HF radio, and whether you have a system that is either a commercial
service or a non-profit setup that is shared with partners.

I am collecting information for the benefit of the East African Internet
Association, Econews Africa and the Southern Africa news consortium MISA
for a workshop in Joburg next week. We in OCHA Nairobi are also
considering whether there might be a role for a public-access non-profit
HF radio e-mail Hub in the region. My findings so far are at:

http://www.unep.org/~parkerb/radio/radio.html

There are a lot of gaps, so please give me your feedback.

Many thanks,

Ben Parker
Deputy Coordinator
UN OCHA
Integrated Regional Information Network
for Central and Eastern Africa

2. Re: HF to Internet connection

From: forrerj@peak.org (Johan Forrer)
Date: Fri, 01 May 1998 20:04:29 -0800
To: ben@ocha.unon.org, hfsig@tapr.org
Message-ID: <199805020306.UAA14454@peak.org>
Rendered body type: text/plain

Ben,

This is in response to your inquiry.

Here is one elegant, low cost, and effective solution that you may consider.
Not always that those features go together:

1) A networked system of interconnected nodes connected by HF radio by AX.25
protocol standard.
2) Each node consists of a minimalist computer, radio modem, and SSB radio.
3) Nodes can be store & forward gateways with internet connections.
4) Gateways may optionally be running Linux -- stable, good performance, and
free.
5) Nodes can be either intelligent, i.e., an Internet gateway, or a user
console for point-to-point communications.
6) Modest hardware requirements: 386/486/Pentium or laptop, DSP-based modem,
SSB radio.
7) Proven DSP modem software -- HF channel simulator results are available
showing superior performance.
    Note: SSB-bandwidth radio channels are required.

The sofware is public domain and the hardware is commonly available.
System engineering/integration could be carried out and performed by radio

technicians -- some computer and radio
skills required. System maintenance is possble by operators with some
aptitude and common sense.

Hope this is of interest. Feel free to get in touch if there are questions.

Johan Forrer

email: forrerj@peak.org
Web page: http://www.peak.org/~forrerj

>Please respond directly to Ben (see below).
>
>Cheers/ Rick W0TN
>-----------------------
>
>From: Ben Parker [SMTP:ben@ocha.unon.org]
>Sent: Tuesday, April 28, 1998 5:37 AM
>To: gwyn@paccomm.com
>Subject: [GATEWAY:182] HF radio e-mail
>
>
>Apologies to those who have already heard from me.
>
>I wonder if you could take the time to let me know what hardware and
>software you are aware of that achieves e-mail (to and from Internet) over
>HF radio, and whether you have a system that is either a commercial
>service or a non-profit setup that is shared with partners.
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>Association, Econews Africa and the Southern Africa news consortium MISA
>for a workshop in Joburg next week. We in OCHA Nairobi are also
>considering whether there might be a role for a public-access non-profit
>HF radio e-mail Hub in the region. My findings so far are at:
>
>http://www.unep.org/~parkerb/radio/radio.html
>
>There are a lot of gaps, so please give me your feedback.
>
>Many thanks,
>
>Ben Parker
>Deputy Coordinator
>UN OCHA
>Integrated Regional Information Network
>for Central and Eastern Africa
>
>

3. Dayton 1998 TAPR Digital Forum and Pack*Bash

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Sat, 2 May 1998 10:54:47 -0500
To: "HF SIG list   mailing"<hfsig@tapr.org>, " Spread Spectrum "<ss@tapr.org>,
        " TAPR/AMSAT DSP "<dsp@tapr.org>,
        "APRS SIG list   mailing"<aprssig@tapr.org>,
        "NETSIG list   mailing"<netsig@tapr.org>,
        "BBS SIG list   mailing"<bbssig@tapr.org>,
        " tacgps "<tacgps@tapr.org>, aprsnews@tapr.org, mic-e@tapr.org,
        TAPR Regional Freq <regional_freq@tapr.org>
Message-ID: <v04003a02b170efeb4e04@[207.8.125.50]>
Rendered body type: text/plain

Dayton 1998 TAPR Digital Forum and PACK*BASH

Dayton Hamvention will be here later this month. Just as a reminder, the
TAPR Digital Forum is on Friday and this year starts much earlier than in
years past. In addition, the PacketBASH banquet will be held on Friday
night.

If you haven't been to Dayton before, you should think about coming this
year. The dates for this year are May 15th - 17th. Check out the web page
http://www.tapr.org/tapr/html/dayton.html for any last minute changes in
the schedule.

The schedule for the TAPR digital forum on Friday is:

10:00 - 10:05am (5 min)
        Introduction to Forum
        John Ackermann, N8UR

10:05 - 10:25am (20 min)
        TAPR Kit/Project/Organizational Update - Greg Jones , WD5IVD
        TAPR Update and discussion on current activities.

10:25 - 10:45am (20 min)
        Regulatory Issues - Dewayne Hendricks, WA8DZP
        Update on Regulatory Issues related to Digital activity.

10:45 - 11:30am (45 min)
        TAPR 900Mhz FHSS Radio Project
        Tom McDermott, N5EG / Greg Jones, WD5IVD
        Latest Information on Status and design of 900Mhz FHSS Radio Design.

11:30 - 12:15pm (45 min)
        GPS Projects - Steve Bible, N7HPR / Tom Clark, W3IWI
        TAPR has introduced several GPS related items in the last year.
        DGPS, GPS Interface boards, TAC-2, Future TOC add-on, etc

12:15 - 12:30pm (15 min)
        Break

12:30 - 1:00pm (30 min)
        Linux AX.25 - John Ackermann, N8UR
        Making AX.25 play on Linux

1:00pm - 1:45pm (60 min)
        APRS SIG and APRS Update
        Stan Horzepa, WA1LOU, APRS SIG Chair with the APRS Authors
        SIG meeting discussing current and future activities in APRS.

PACK*BASH and TAPR Banquet

Attending the 1998 Dayton Hamvention? Then don't miss out on PACK*BASH and
TAPR Banquet.

What?
        An event for the digitally-inclined ham, featuring:
                * Buffet dinner
                * Speaker Doug McKinney, KC3RL
                      discussing GPS and his Bouy projects in the Pacific
                * TAPR special interest group meetings
                * Prize Drawing!

When?
        Friday evening, May 15, 1998
        Doors open at 7:00 pm; dinner served at 7:30 pm
        Speaker and meetings after dinner

Where?
        NCR "Sugarcamp" Conference Center, 101 W. Schantz Ave. --
        South of downtown Dayton, just west of Main St. (Rt 48).
        Maps available on the TAPR web site or at the TAPR booth.
        (http://www.tapr.org/tapr/html/dayton.html)

How?
        Dinner requires advance registration and payment through TAPR.
        Tickets will be available at the TAPR booth on Friday, though
        we strongly encourage registration <before> Hamvention. The cost
        is $20.00 per person, tax and tip included.

        All amateurs are welcome to attend, enjoy the speaker, and
        participate in the meetings, although only those purchasing a
        dinner can eat.

        To register, contact:
                PACK*BASH

c/o TAPR
8987-309 E. Tanque Verde Road #337
Tucson, AZ 85749-9399

Phone: (940) 383-0000
Fax: (940) 566-2544
Internet: tapr@tapr.org

Visa/Mastercard Accepted

Who?
PACK*BASH is co-sponsored by TAPR -- Tucson Amateur Packet
Radio, the national leader in digital communication -- and the
Miami Valley FM Association, Dayton's packet radio club.

For more information (including maps), go to
http://www.tapr.org/tapr/html/dayton.html, or send e-mail to
tapr@tapr.org. You can request information via packet radio at
BASH@N8ACV.#DAY.OH.USA.NOAM.

Tucson Amateur Packet Radio
8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399
---
e-mail: tapr@tapr.org web: <http://www.tapr.org/>
ftp: ftp.tapr.org
phone: 940-383-0000 fax: 940-565-2544
---

4. Re: HF to Internet connection

From: Ben Parker <ben@ocha.unon.org>
Date: Sat, 2 May 1998 17:52:35 -0300 (GMT+3)
To: Johan Forrer <forrerj@peak.org>
Cc: hfsig@tapr.org
Message-ID: <Pine.LNX.3.95.980502175108.7480B-100000@amahoro.dha.unon.org>
Rendered body type: text/plain

Indeed. AX.25 and KA9Q - type solutions I will have to do my homework on.
I will get back to you with question once I even half know what I'm
talking about.

Many thanks for the feedback.

Ben

On Fri, 1 May 1998, Johan Forrer wrote:

> Ben,
>
> This is in response to your inquiry.
>
> Here is one elegant, low cost, and effective solution that you may consider.
> Not always that those features go together:
>
> 1) A networked system of interconnected nodes connected by HF radio by AX.25
> protocol standard.
> 2) Each node consists of a minimalist computer, radio modem, and SSB radio.
> 3) Nodes can be store & forward gateways with internet connections.
> 4) Gateways may optionally be running Linux -- stable, good performance, and
> free.
> 5) Nodes can be either intelligent, i.e., an Internet gateway, or a user
> console for point-to-point communications.
> 6) Modest hardware requirements: 386/486/Pentium or laptop, DSP-based modem,
> SSB radio.
> 7) Proven DSP modem software -- HF channel simulator results are available
> showing superior performance.
>   Note: SSB-bandwidth radio channels are required.
>
> The sofware is public domain and the hardware is commonly available.
> System engineering/integration could be carried out and performed by radio
> technicians -- some computer and radio
> skills required. System maintenance is possble by operators with some
> aptitude and common sense.
>
> Hope this is of interest. Feel free to get in touch if there are questions.

>
> Johan Forrer
>
> email: forrerj@peak.org
> Web page: http://www.peak.org/~forrerj
> --
> ----------------------------
>
>
>
>
>
> >Please respond directly to Ben (see below).
> >
> >Cheers/ Rick W0TN
> >-----------------------
> >
> >From: Ben Parker [SMTP:ben@ocha.unon.org]
> >Sent: Tuesday, April 28, 1998 5:37 AM
> >To: gwyn@paccomm.com
> >Subject: [GATEWAY:182] HF radio e-mail
> >
> >
> >Apologies to those who have already heard from me.
> >
> >I wonder if you could take the time to let me know what hardware and
> >software you are aware of that achieves e-mail (to and from Internet) over
> >HF radio, and whether you have a system that is either a commercial
> >service or a non-profit setup that is shared with partners.
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> >I am collecting information for the benefit of the East African Internet
> >Association, Econews Africa and the Southern Africa news consortium MISA
> >for a workshop in Joburg next week. We in OCHA Nairobi are also
> >considering whether there might be a role for a public-access non-profit
> >HF radio e-mail Hub in the region. My findings so far are at:
> >
> >http://www.unep.org/~parkerb/radio/radio.html
> >
> >There are a lot of gaps, so please give me your feedback.
> >
> >Many thanks,
> >
> >Ben Parker
> >Deputy Coordinator
> >UN OCHA
> >Integrated Regional Information Network
> >for Central and Eastern Africa
> >
> >
>

5. Computing Eb/N0 for simulated digital signals ?

From: Dquagliana <Dquagliana@aol.com>
Date: Sun, 3 May 1998 23:03:32 EDT
To: hfsig@tapr.org
Message-ID: <b68d0b39.354d3005@aol.com>
Rendered body type: text/plain

I know how to digitally generate my signal, and I can
also generate samples of AWGN.

How can I add the two together in such a way
as to generate a simulated signal with noise
at a given Eb/N0 ?

Stated another way, what do I need to know about
the white noise samples to calculate the Eb/N0
when they are added to my signal samples ?

Doug KA2UPW
dquagliana@aol.com

6. Re: [HFSIG:2677] Computing Eb/N0 for simulated digital signals ?

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 04 May 1998 21:16:17 -0800
To: hfsig@tapr.org
Message-ID: <199805050418.VAA26874@peak.org>
Rendered body type: text/plain

>I know how to digitally generate my signal, and I can
>also generate samples of AWGN.
>
>How can I add the two together in such a way
>as to generate a simulated signal with noise
>at a given Eb/N0 ?
>
>Stated another way, what do I need to know about
>the white noise samples to calculate the Eb/N0
>when they are added to my signal samples ?
>
>Doug KA2UPW
>dquagliana@aol.com
>

Doug,

There are some useful materials in Tom McDermott's book: Wireless Digital Communications:
Design and Theory (page 25) on Eb/No calculations (see TAPR web page about the book).

You may also derive Eb/No from the definition of SNR = 10 log10(signal power/noise power),
and how SNR is related to Eb/No. This requires that you take noise bandwidth and signaling rate
into account. Be a bit careful of how signal power gets derived, and precisely what the definition
of a "bit" means in this context otherwise things may not appear what you think.

Easier said than done!

You may also search the HFSIG archives for discussions that we have had on this topic.

Maybe someone else may share some ideas?

73's

--Johan, KC7WW

7. Remove

From: "oris" <oris@dirtball.com>
Date: Tue, 5 May 1998 23:32:24 -0400
To: <hfsig@tapr.org>
Message-ID: <199805060332.WAA21594@client.alltel.net>
Rendered body type: text/plain

http://www.dirtball.com
oris@dirtball.com
N3CND@N3CND.#NEGA.GA.USA.NOAM
I did not request your messages please remove me

> From: hfsig@tapr.org
> To: hfsig@tapr.org
> Subject: [HFSIG:2678] HFSIG digest 503
> Date: Tuesday, May 05, 1998 8:39 PM
>
>   HFSIG Digest 503
>
> Topics covered in this issue include:
>
>   1) Re: Computing Eb/N0 for simulated digital signals ?
>   by forrerj@peak.org (Johan Forrer)
>
> --
>
> Date: Mon, 04 May 1998 21:16:17 -0800
> From: forrerj@peak.org (Johan Forrer)
> To: hfsig@tapr.org
> Subject: Re: Computing Eb/N0 for simulated digital signals ?
> Message-ID: <199805050418.VAA26874@peak.org>
>
> >I know how to digitally generate my signal, and I can
> >also generate samples of AWGN.
> >
> >How can I add the two together in such a way

> >as to generate a simulated signal with noise
> >at a given Eb/N0 ?
> >
> >Stated another way, what do I need to know about
> >the white noise samples to calculate the Eb/N0
> >when they are added to my signal samples ?
> >
> >Doug KA2UPW
> >dquagliana@aol.com
> >
>
>
> Doug,
>
> There are some useful materials in Tom McDermott's book: Wireless Digital
> Communications:
> Design and Theory (page 25) on Eb/No calculations (see TAPR web page
about
> the book).
>
> You may also derive Eb/No from the definition of SNR = 10 log10(signal
> power/noise power),
> and how SNR is related to Eb/No. This requires that you take noise
bandwidth
> and signaling rate
> into account. Be a bit careful of how signal power gets derived, and
> precisely what the definition
> of a "bit" means in this context otherwise things may not appear what you
> think.
>
> Easier said than done!
>
> You may also search the HFSIG archives for discussions that we have had
on
> this topic.
>
> Maybe someone else may share some ideas?
>
> 73's
>
> --Johan, KC7WW
>
>
> ------------------------------
>
> End of HFSIG Digest 503
> **********************

8. TEST

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Thu, 7 May 1998 00:47:51 -0500
To: "HF SIG list   mailing"<hfsig@tapr.org>
Message-ID: <v04003a0fb176fb7eb106@[207.8.125.50]>
Rendered body type: text/plain

TEST

Greg

Greg Jones, WD5IVD   Austin, Texas
wd5ivd@tapr.org   http://www.tapr.org/~wd5ivd

Real World, The n.:
        5. Anywhere outside a university. "Poor fellow, he's left UT
Austin and gone into the real world."

9. TEST

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Thu, 7 May 1998 18:52:32 -0500
To: "HF SIG list   mailing"<hfsig@tapr.org>
Message-ID: <v04003a30b177f9b28388@[207.8.125.50]>
Rendered body type: text/plain

TEST

We had a little problem with the HFSIG.

Testing it.

Greg

Greg Jones, WD5IVD Austin, Texas
wd5ivd@tapr.org http://www.tapr.org/~wd5ivd

Real World, The n.:
5. Anywhere outside a university. "Poor fellow, he's left UT
Austin and gone into the real world."

10. TEST 2

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Thu, 7 May 1998 19:44:08 -0500
To: "HF SIG list mailing"<hfsig@tapr.org>
Message-ID: <v04003a33b17805cf5d8d@[207.8.125.50]>
Rendered body type: text/plain

This is a second test.

Greg

Greg Jones, WD5IVD Austin, Texas
wd5ivd@tapr.org http://www.tapr.org/~wd5ivd

Real World, The n.:
5. Anywhere outside a university. "Poor fellow, he's left UT
Austin and gone into the real world."

11. Re: [HFSIG:2682] TEST 2

From: k5cnf1@juno.com (Richard B Morrow)
Date: Thu, 7 May 1998 21:09:06 -0500
To: hfsig@tapr.org
Message-ID: <19980507.210906.3550.3.k5cnf1@juno.com>
Rendered body type: text/plain

This one also got here
K5CNF

On Thu, 7 May 1998 19:48:34 -0500 (CDT) "Greg Jones, WD5IVD"
<wd5ivd@tapr.org> writes:
>This is a second test.
>
>Greg
>
>-----
>Greg Jones, WD5IVD Austin, Texas
>wd5ivd@tapr.org http://www.tapr.org/~wd5ivd
>
>
>Real World, The n.:
> 5. Anywhere outside a university. "Poor fellow, he's left UT
>Austin and gone into the real world."
>
>
>

_____
You don't need to buy Internet access to use free Internet e-mail.
Get completely free e-mail from Juno at http://www.juno.com
Or call Juno at (800) 654-JUNO [654-5866]

12. Re: [HFSIG:2681] TEST

From: k5cnf1@juno.com (Richard B Morrow)
Date: Thu, 7 May 1998 21:08:31 -0500
To: hfsig@tapr.org
Message-ID: <19980507.210906.3550.2.k5cnf1@juno.com>
Rendered body type: text/plain

Got to me, wondered what had happened.
K5CNF

On Thu, 7 May 1998 18:57:39 -0500 (CDT) "Greg Jones, WD5IVD"
<wd5ivd@tapr.org> writes:

>TEST
>
>We had a little problem with the HFSIG.
>
>Testing it.
>
>Greg
>
>-----
>Greg Jones, WD5IVD   Austin, Texas
>wd5ivd@tapr.org   http://www.tapr.org/~wd5ivd
>
>
>Real World, The n.:
>   5. Anywhere outside a university. "Poor fellow, he's left UT
>Austin and gone into the real world."
>
>
>

_____
You don't need to buy Internet access to use free Internet e-mail.
Get completely free e-mail from Juno at http://www.juno.com
Or call Juno at (800) 654-JUNO [654-5866]

13. Re: [HFSIG:2683] Re: TEST 2

From: CDrentea <CDrentea@aol.com>
Date: Fri, 8 May 1998 00:49:00 EDT
To: hfsig@tapr.org
Message-ID: <1aff524f.35528ebd@aol.com>
Rendered body type: text/plain

Why am I getting all these tests? Am I supose to know you?

73, Cornell WB3JZO

14. One last Test

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Thu, 7 May 1998 23:55:46 -0500
To: "HF SIG list   mailing"<hfsig@tapr.org>
Message-ID: <v04003a3ab17840923546@[207.8.125.50]>
Rendered body type: text/plain

I think I have everything back to normal.

Sorry for the additional messages.

Greg

Greg Jones, WD5IVD   Austin, Texas
wd5ivd@tapr.org   http://www.tapr.org/~wd5ivd

Real World, The n.:
        5. Anywhere outside a university. "Poor fellow, he's left UT
Austin and gone into the real world."

15. Turbo codes on Rayleigh channels

From: Tom McDermott <mcdermot@aud.alcatel.com>
Date: Fri, 8 May 1998 12:45:13 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BD7A7F.25429E00.mcdermot@aud.alcatel.com>
Rendered body type: text/plain

This is a very good article recently published: "Design and Analysis
of Turbo Codes on Ralyeigh Fading Channels" by Eric Hall & Stephen Wilson,
IEEE Journal on Selected Areas in Communications, Vol 16 No 2,
Feb 98, pp 160-174

For interleaving factors of 500, the performance is not that good, but
for interleaving factors of 5,000 and up, the BER performance in
simulations is very good indeed for a rate=1/3 coder (within 1.5 dB of
of AWGN channel and 50,000 interleaver, no side information, BTs=0.1).
The interleavers are fixed.

As the fading process slows (BTs <= .01), the performance deteriorates.

Tom McDermott
mcdermot@aud.alcatel.com

16. Re: [HFSIG:2686] One last Test

From: Gary Chatters <gc@cen.com>
Date: Fri, 8 May 1998 14:08:15 -0400 (EDT)
To: hfsig@tapr.org
Cc: wd5ivd@tapr.org
Message-ID: <199805081808.OAA07139@fox.cen.com>
Rendered body type: text/plain

>
>I think I have everything back to normal.
>
>Sorry for the additional messages.
>
>Greg
>

Could you give us a little more explanation?

About 3 months ago I noticed that I got dropped from the list
and so did most every one else. By the time I resubscribed there
were only about a dozen subscribers.

When I resubscribed it was from a different address. Now the old
address is getting HFSIG mail again, starting with your first
test message.

TNX ES 73,

Gary

17. Re: [HFSIG:2686] One last Test

From: Monte.Tremont@nextel.com (Monte Tremont)
Date: Sun, 10 May 1998 12:52:04 -0400
To: hfsig@tapr.org, "Greg Jones; WD5IVD" <wd5ivd@tapr.org>
Message-ID: <000371AF.CE21050@nextel.com>
Rendered body type: text/plain

No its not fixed if I'm still getting these messages >>>>>>>>>>>>>

     If you wish to discuss this further please contact me at
     wb2eyf@erols.com and not AT MY EMPLOYERS EMAIL ADDRESS Thank
     you.>>>>>>>>>>>>>>>>

_____ Reply Separator _____
Subject: [HFSIG:2686] One last Test
Author: "Greg Jones; WD5IVD" <wd5ivd@tapr.org> at INTERNET
Date: 5/8/1998 12:01 AM

I think I have everything back to normal.

Sorry for the additional messages.

Greg

Greg Jones, WD5IVD   Austin, Texas
wd5ivd@tapr.org   http://www.tapr.org/~wd5ivd

Real World, The n.:
        5. Anywhere outside a university. "Poor fellow, he's left UT
Austin and gone into the real world."

18. Re: [HFSIG:2688] Re: One last Test

From: Monte.Tremont@nextel.com (Monte Tremont)
Date: Sun, 10 May 1998 12:59:20 -0400
To: hfsig@tapr.org, Gary Chatters <gc@cen.com>
Message-ID: <000371B1.CE21050@nextel.com>
Rendered body type: text/plain

Nice story .... now delete me from this "old" but now CURRENT listing.
     Thanks The MIS dept caught this one and I hope I've covered my tail by
     showing what a bunch of %^$#-ups the HF sig gate keeper is! 73 I

think ...???!!!

_____ Reply Separator _____
Subject: [HFSIG:2688] Re: One last Test
Author: Gary Chatters <gc@cen.com> at INTERNET
Date: 5/8/1998 1:10 PM

>
>I think I have everything back to normal.
>
>Sorry for the additional messages.
>
>Greg
>

Could you give us a little more explanation?

About 3 months ago I noticed that I got dropped from the list
and so did most every one else. By the time I resubscribed there
were only about a dozen subscribers.

When I resubscribed it was from a different address. Now the old
address is getting HFSIG mail again, starting with your first
test message.

TNX ES 73,

Gary

19. IPR-2000 TCP/IP via HF-Radio

From: gateway@vita.org (by way of The Whiting Household <rwhiting@winternet.com>)
Date: Mon, 18 May 1998 18:47:49 -0500
To: hfsig@tapr.org, tlan@twinslan.org
Cc: scott.e.morgan@LMCO.com (Scott Morgan N6RVH)
Message-ID: <3.0.5.32.19980518184749.007cf590@mail.winternet.com>
Rendered body type: text/plain

Thought HF data fans might be interested in the product describrd below.

Cheers/ Rick W0TN

GATEWAY Digest 94

Topics covered in this issue include:

  1) EURAF announces the IPR-2000 TCP/IP via HF-Radio !
       by Peter Schulze <dts@dtsdata.intnet.bj>

Topic No. 1

Date: Mon, 18 May 1998 12:51:40 +-200
From: Peter Schulze <dts@dtsdata.intnet.bj>
To: "'gateway@vita.org'" <gateway@vita.org>
Subject: EURAF announces the IPR-2000 TCP/IP via HF-Radio !
Message-ID: <01BD825B.B46CB5A0@dtsdata.intnet.bj>

EURAF is pleased to announce its latest HF radio product: the IPR-2000.

The IPR-2000 is the first Internet/Intranet access router for HF-radio =
links. Offering unprecedented
flexibility and functionality, the IPR-2000 router connects remote PC's =
or remote offices with Ethernet
LANs to the Internet and to company Intranets.=20

Using EURAF's unique TCP/IP via radio technology you get automated and =
transparent TCP/IP
connectivity between any two points in the world using HF-radio.

Using the latest HF modem technology and highly efficient data =
compression algorithms the IPR-2000 assures fast connections. Further, =
the IPR-2000 applies a special buffering and spoofing technology that =
greatly enhances throughput. EURAF's unique protocol technology removes =
most of the IP protocol overhead while the data is sent across the radio =
link.
The protocol also takes the special timing requirements of the HF modems =

into account making the system extremely efficient.=20

Access your E-mail, browse the Web, get files via FTP, access servers =
via Telnet, IPR-2000 will do it!

Please visit our Website at http://dtsdata.intnet.bj for more =
information or contact us at dts@dtsdata.intnet.bj

Best Regards

Peter Schulze

EURAF Cotonou
dts@dtsdata.intnet.bj
http://dtsdata.intnet.bj

End of GATEWAY Digest 94
****

20. Re: [HFSIG:2691] IPR-2000 TCP/IP via HF-Radio

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Tue, 19 May 1998 08:54:42 -0500 (CDT)
To: hfsig@tapr.org, qst@arrl.org
Cc: kj5ae@arrl.org, w5zn@arrl.org, w2ml@arrl.org, w6rod@arrl.org,
w5jbp@arrl.org, k1zz@arrl.org, wb8imy@arrl.org, wd5ivd@tapr.org
Message-ID: <Pine.SOL.3.91.980519080033.544A-100000@www.kelly-afb.org>
Rendered body type: text/plain

Strange...but I've known that something like this would surface at some
point in time.

I've great read with interest Dave Sumner's editorials in QST over the
last two years calling for "new" technology in the ham community...but I
can't recall seeing Dave or Steve Ford call for specific articles on
specific topics. Please correct me if I'm wrong.

There are a number of highly educated, _brilliant_, individuals who
"monitor" and regularly contribute to this mail reflector, who have the
capability of putting their heads together to come up with an article(s)
telling us, "appliance" operators, what hardware (TAPR/HAL/?) and software
to buy/obtain that would allow the run-of-the-mill ham to run high speed
HF data (E-Mail if you please and near or over 2400 BPS) using our
ordinary "rice-box" HF rig...and moreover our VHF/UHF rigs and with little
or no modification.

And how about the ARRL calling for specific article topics and ICOM and
MCM Electronics (the ham electronics manufacturers) to build PC controlled
transceivers and/or PC card transceivers...certainly the technology is
there. Look at what KACHINA, MCM and ICOM have done already. How about a
QST report on the functional capabilities of the KACHINA, MCM WinRadio and
the ICOM PC Radio and slip in a call for MCM and ICOM in include
transmitter sections?

The corporate knowledge to accomplish the above resides within the
audience of this E-Mail. The technical capability resides within those who
monitor the HFSIG E-Mail reflector. The ARRL _HAS_ the capability to place
it before the U.S. Amateur Radio Operator and _LEAD_ the effort.

Will you, our technical and corporate leadership, make the call?
Please act now.

Best Regards,

Walt DuBose/K5YFW

==========

On Mon, 18 May 1998 gateway@vita.org wrote:

> Thought HF data fans might be interested in the product describrd below.
>
> Cheers/ Rick W0TN
> ---
>   GATEWAY Digest 94
>
> Topics covered in this issue include:

>
>   1) EURAF announces the IPR-2000 TCP/IP via HF-Radio !
>   by Peter Schulze <dts@dtsdata.intnet.bj>
>
> --
>
> Topic No. 1
>
> Date: Mon, 18 May 1998 12:51:40 +-200
> From: Peter Schulze <dts@dtsdata.intnet.bj>
> To: "'gateway@vita.org'" <gateway@vita.org>
> Subject: EURAF announces the IPR-2000 TCP/IP via HF-Radio !
> Message-ID: <01BD825B.B46CB5A0@dtsdata.intnet.bj>
>
> EURAF is pleased to announce its latest HF radio product: the IPR-2000.
>
> The IPR-2000 is the first Internet/Intranet access router for HF-radio =
> links. Offering unprecedented
> flexibility and functionality, the IPR-2000 router connects remote PC's =
> or remote offices with Ethernet
> LANs to the Internet and to company Intranets.=20
>
> Using EURAF's unique TCP/IP via radio technology you get automated and =
> transparent TCP/IP
> connectivity between any two points in the world using HF-radio.
>
> Using the latest HF modem technology and highly efficient data =
> compression algorithms the IPR-2000 assures fast connections. Further, =
> the IPR-2000 applies a special buffering and spoofing technology that =
> greatly enhances throughput. EURAF's unique protocol technology removes =
> most of the IP protocol overhead while the data is sent across the radio =
> link.
> The protocol also takes the special timing requirements of the HF modems =
> into account making the system extremely efficient.=20
>
> Access your E-mail, browse the Web, get files via FTP, access servers =
> via Telnet, IPR-2000 will do it!
>
> Please visit our Website at http://dtsdata.intnet.bj for more =
> information or contact us at dts@dtsdata.intnet.bj
>
> Best Regards
>
> Peter Schulze
>
> EURAF Cotonou
> dts@dtsdata.intnet.bj
> http://dtsdata.intnet.bj
>
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>
> End of GATEWAY Digest 94
> ************************
>
>
>

21. Re: [HFSIG:2692] Re: IPR-2000 TCP/IP via HF-Radio

From: "B.HAINEAULT" <bhaineau@cts.com>
Date: Tue, 19 May 1998 19:03:50 -0700
To: <hfsig@tapr.org>
Message-ID: <001e01bd8393$893ab880$d805a3cd@bhaineau.cts.com>
Rendered body type: text/plain

Hi Walt,

The technology exists... HAL's DSP-4100/2k "CLOVER-2000" modem
(http://www.halcomm.com), whips along at speeds of up to 3000bps... even
faster perceived throughputs if you include compression. The problem is that
the FCC won't let us use this modem on ham bands as it requires 2kHz
bandwidth. As you know, the FCC limits data bandwidths on HF to 1kHz.   This
limitations does not exist in many other countries.

Up to know, we have been using a variant of this waveform, called X-CLOVER,
on experimental frequencies across the HF spectrum (for example, listen
sometimes on 16.4036 MHz, USB, for high-speed fax/data tests). This
waveform is being used by various diplomatic networks worldwide. HF is far
from dead... it's in a transition phase from voice to efficient and

automated data exchanges. We use an adaptation of a program written by EURAF (called DTS). It was based on the amateur offering called "Express for CLOVER". The creators are now running a lucrative "commercial" business customizing HF network software and TCP/IP-HF systems.

There are also MIL-STD-110A modems that have been around for many years. They run at 4800-2400bps/PSK but their cost is prohibitively high. As part of the HFIA (NTIA, etc.), we are working on 4800 and 9600bps HF modems (proposed "64QAM-based" MIL-STD-110B)... but they require around 4kHz and cost a fortune ;-) today.

The HAL "CLOVER-2000" modems are affordable and very efficient... SCS also has a very powerful PACTOR-II modem running at up to 850bps (+compression).

Now if we can only get the FCC to loosen-up their 19th-century regulations ;-) and let hams do one of the things they are supposed to do: " experiment new modes".

Ham applications ?

High-speed nationwide HF traffic systems... running at enormously higher efficiency levels than AX.25 packet over HF... and many more.

73 de Bruno Haineault, KF6QDP (and experimental license holder for DATRON)
San Marcos, California

ICQ Pager #: 335336
ICQ Pager address: http://wwp.mirabilis.com/335336

-----Original Message-----
From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
To: hfsig@tapr.org <hfsig@tapr.org>
Date: Tuesday, 1998 May, 19 07:44
Subject: [HFSIG:2692] Re: IPR-2000 TCP/IP via HF-Radio

>Strange...but I've known that something like this would surface at some
>point in time.
>
>I've great read with interest Dave Sumner's editorials in QST over the
>last two years calling for "new" technology in the ham community...but I
>can't recall seeing Dave or Steve Ford call for specific articles on
>specific topics. Please correct me if I'm wrong.
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>There are a number of highly educated, _brilliant_, individuals who
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>capability of putting their heads together to come up with an article(s)
>telling us, "appliance" operators, what hardware (TAPR/HAL/?) and software
>to buy/obtain that would allow the run-of-the-mill ham to run high speed
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>or no modification.
>
>And how about the ARRL calling for specific article topics and ICOM and
>MCM Electronics (the ham electronics manufacturers) to build PC controlled
>transceivers and/or PC card transceivers...certainly the technology is
>there. Look at what KACHINA, MCM and ICOM have done already. How about a
>QST report on the functional capabilities of the KACHINA, MCM WinRadio and
>the ICOM PC Radio and slip in a call for MCM and ICOM in include
>transmitter sections?
>
>The corporate knowledge to accomplish the above resides within the
>audience of this E-Mail. The technical capability resides within those who
>monitor the HFSIG E-Mail reflector. The ARRL _HAS_ the capability to place
>it before the U.S. Amateur Radio Operator and _LEAD_ the effort.
>
>Will you, our technical and corporate leadership, make the call?
>_Please act now._
>
>Best Regards,
>
>Walt DuBose/K5YFW
>
>==========
==
>
>On Mon, 18 May 1998 gateway@vita.org wrote:
>
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>>
>> Cheers/ Rick W0TN
>> ---
>>   GATEWAY Digest 94
>>
>> Topics covered in this issue include:
>>
>>   1) EURAF announces the IPR-2000 TCP/IP via HF-Radio !
>> by Peter Schulze <dts@dtsdata.intnet.bj>
>>
>> --
>>
>> Topic No. 1
>>
>> Date: Mon, 18 May 1998 12:51:40 +-200
>> From: Peter Schulze <dts@dtsdata.intnet.bj>
>> To: "'gateway@vita.org'" <gateway@vita.org>
>> Subject: EURAF announces the IPR-2000 TCP/IP via HF-Radio !
>> Message-ID: <01BD825B.B46CB5A0@dtsdata.intnet.bj>
>>
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>> The IPR-2000 is the first Internet/Intranet access router for HF-radio =
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>> flexibility and functionality, the IPR-2000 router connects remote PC's =
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>>
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>> most of the IP protocol overhead while the data is sent across the radio
=
>> link.
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=
>> into account making the system extremely efficient.=20
>>
>> Access your E-mail, browse the Web, get files via FTP, access servers =
>> via Telnet, IPR-2000 will do it!
>>
>> Please visit our Website at http://dtsdata.intnet.bj for more =
>> information or contact us at dts@dtsdata.intnet.bj
>>
>> Best Regards
>>
>> Peter Schulze
>>
>> EURAF Cotonou
>> dts@dtsdata.intnet.bj
>> http://dtsdata.intnet.bj
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>> End of GATEWAY Digest 94
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22. Re: [HFSIG:2693] Re: IPR-2000 TCP/IP via HF-Radio

From: "Louis Cobet" <louis_cobet@email.msn.com>
Date: Tue, 19 May 1998 22:38:24 -0700
To: <hfsig@tapr.org>
Message-ID: <0142f3542051458UPIMSSMTPUSR05@email.msn.com>
Rendered body type: text/plain

Bruno,

My understanding is that 1200 bps is legal on 10 meters. With propagation

improving this could be a useful point of experimentation.

Louis Cobet, K6MDH
louis_cobet@email.msn.com

You wrote:

>Hi Walt,
>
>The technology exists... HAL's DSP-4100/2k "CLOVER-2000" modem
>(http://www.halcomm.com), whips along at speeds of up to 3000bps... even
>faster perceived throughputs if you include compression. The problem is
that
>the FCC won't let us use this modem on ham bands as it requires 2kHz
>bandwidth. As you know, the FCC limits data bandwidths on HF to 1kHz.
This
>limitations does not exist in many other countries.
>
>Up to know, we have been using a variant of this waveform, called X-CLOVER,
>on experimental frequencies across the HF spectrum (for example, listen
>sometimes on 16.4036 MHz, USB, for high-speed fax/data tests). This
>waveform is being used by various diplomatic networks worldwide. HF is far
>from dead... it's in a transition phase from voice to efficient and
>automated data exchanges. We use an adaptation of a program written by
EURAF
>(called DTS). It was based on the amateur offering called "Express for
>CLOVER". The creators are now running a lucrative "commercial" business
>customizing HF network software and TCP/IP-HF systems.
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>There are also MIL-STD-110A modems that have been around for many years.
>They run at 4800-2400bps/PSK but their cost is prohibitively high. As part
>of the HFIA (NTIA, etc.), we are working on 4800 and 9600bps HF modems
>(proposed "64QAM-based" MIL-STD-110B)... but they require around 4kHz and
>cost a fortune ;-) today.
>
>The HAL "CLOVER-2000" modems are affordable and very efficient... SCS also
>has a very powerful PACTOR-II modem running at up to 850bps (+compression).
>
>Now if we can only get the FCC to loosen-up their 19th-century regulations
>;-) and let hams do one of the things they are supposed to do: " experiment
>new modes".
>
>Ham applications ?
>
>High-speed nationwide HF traffic systems... running at enormously higher
>efficiency levels than AX.25 packet over HF... and many more.
>
>73 de Bruno Haineault, KF6QDP (and experimental license holder for DATRON)
>San Marcos, California
>
>ICQ Pager #: 335336
>ICQ Pager address: http://wwp.mirabilis.com/335336
>

23. Re: [HFSIG:2694] Re: IPR-2000 TCP/IP via HF-Radio

From: "R. & S. Keller" <kellerrs@dpnet.net>
Date: Wed, 20 May 1998 17:58:28 -0400
To: hfsig@tapr.org
Message-ID: <3.0.3.32.19980520175828.00699a78@zeus.dpnet.net>
Rendered body type: text/plain

REMOVE MY ADDRESS FROM YOUR DISTRIBUTION LIST NOW!
At 12:51 AM 5/20/98 -0500, you wrote:
>Bruno,
>
>
>My understanding is that 1200 bps is legal on 10 meters. With propagation
>improving this could be a useful point of experimentation.
>
>Louis Cobet, K6MDH
>louis_cobet@email.msn.com
>
>You wrote:
>
>>Hi Walt,
>>
>>The technology exists... HAL's DSP-4100/2k "CLOVER-2000" modem
>>(http://www.halcomm.com), whips along at speeds of up to 3000bps... even
>>faster perceived throughputs if you include compression. The problem is

>that
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>>customizing HF network software and TCP/IP-HF systems.
>>
>>There are also MIL-STD-110A modems that have been around for many years.
>>They run at 4800-2400bps/PSK but their cost is prohibitively high. As part
>>of the HFIA (NTIA, etc.), we are working on 4800 and 9600bps HF modems
>>(proposed "64QAM-based" MIL-STD-110B)... but they require around 4kHz and
>>cost a fortune ;-) today.
>>
>>The HAL "CLOVER-2000" modems are affordable and very efficient... SCS also
>>has a very powerful PACTOR-II modem running at up to 850bps (+compression).
>>
>>Now if we can only get the FCC to loosen-up their 19th-century regulations
>>;-) and let hams do one of the things they are supposed to do: " experiment
>>new modes".
>>
>>Ham applications ?
>>
>>High-speed nationwide HF traffic systems... running at enormously higher
>>efficiency levels than AX.25 packet over HF... and many more.
>>
>>73 de Bruno Haineault, KF6QDP (and experimental license holder for DATRON)
>>San Marcos, California
>>
>>ICQ Pager #: 335336
>>ICQ Pager address: http://wwp.mirabilis.com/335336
>>
>
>
>
>
>

24. Re: [HFSIG:2694] Re: IPR-2000 TCP/IP via HF-Radio

From: "B.HAINEAULT" <bhaineau@cts.com>
Date: Wed, 20 May 1998 19:04:04 -0700
To: "Louis Cobet" <louis_cobet@email.msn.com>
Cc: <hfsig@tapr.org>
Message-ID: <000c01bd845c$bc522680$d805a3cd@bhaineau.cts.com>
Rendered body type: text/plain

Hi Louis,

I must admit that I didn't really consider 10m as "HF" ;-) but you are
right. With the improvement in propagation, 10m can be interesting to say
the least.

However, my main point was that we needed less FCC involvement and more ham
experimenting on HF. The high-speed HF data networks would have to work on
other bands as they need to function, even during solar minimums ;-)

73 de Bruno Haineault, KF6QDP
San Marcos, California

ICQ Pager #: 335336
ICQ Pager address: http://wwp.mirabilis.com/335336

-----Original Message-----
From: Louis Cobet <louis_cobet@email.msn.com>
To: hfsig@tapr.org <hfsig@tapr.org>
Date: Tuesday, 1998 May, 19 23:17
Subject: [HFSIG:2694] Re: IPR-2000 TCP/IP via HF-Radio

>Bruno,

>
>
>My understanding is that 1200 bps is legal on 10 meters. With propagation
>improving this could be a useful point of experimentation.
>
>Louis Cobet, K6MDH
>louis_cobet@email.msn.com

25. Re: [HFSIG:2696] Re: IPR-2000 TCP/IP via HF-Radio

From: dona@tridelta.com (Don Adams)
Date: Thu, 21 May 1998 08:18:30 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199805211218.IAA19134@sparcy.tridelta.com>
Rendered body type: text/plain

>
>>My understanding is that 1200 bps is legal on 10 meters.

Is that 1200 bps or 1200 baud?

26. Re: [HFSIG:2693] Re: IPR-2000 TCP/IP via HF-Radio

From: Anthony Martin <anthony.martin@gecm.com>
Date: Thu, 21 May 1998 18:17:42 +0100
To: hfsig@tapr.org
Message-ID: <356461B6.5EC4@gecm.com>
Rendered body type: text/plain

B.HAINEAULT wrote:
>
> Hi Walt,
>
> The technology exists... HAL's DSP-4100/2k "CLOVER-2000" modem
> (http://www.halcomm.com), whips along at speeds of up to 3000bps... even
> faster perceived throughputs if you include compression. The problem is that
> the FCC won't let us use this modem on ham bands as it requires 2kHz
> bandwidth. As you know, the FCC limits data bandwidths on HF to 1kHz.   This
> limitations does not exist in many other countries.
>
How many channels of 1kHz are you allowed to occupy simultaneously?
What if you put them side-by-side? What if you then put them thru
the same IF, RF & antenna? (Stoopid rule, what?)

If you digitise voice and send it using digital modulation, is that
voice and allowed 3kHz?

>
> Now if we can only get the FCC to loosen-up their 19th-century regulations
> ;-) and let hams do one of the things they are supposed to do: " experiment
> new modes".

Here here, that archological compulsory digital mode `morse' will soon
be gone too so then I will join the licenced hams working on HF
internet...

Ant.
--
_____

A.N.Martin   E-mail anthony.martin@gecm.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
_____

27. Test

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.tu-darmstadt.de>
Date: Fri, 29 May 1998 09:53:00 +0200
To: hfsig@tapr.org
Message-ID: <356E695C.15B06F40@zeus.uet.e-technik.tu-darmstadt.de>
Rendered body type: text/plain

Hi!

Is there really no activity going on for months?

73' Alexander

--
------------------------------+--------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| technik u. Elektroakustik | Fax : +49-6151-165545
|
| Merckstrasse 25 | EMail: a.kurpiers@uet.tu-darmstadt.de
|
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
------------------------------+--------------------------------------

28. TAPR Office Closure

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Sat, 30 May 1998 13:42:11 -0500
To: "TAPR-BB list   mailing"<tapr-bb@tapr.org>,
        "NETSIG list   mailing"<netsig@tapr.org>,
        "APRS SIG list   mailing"<aprssig@tapr.org>,
        "HF SIG list   mailing"<hfsig@tapr.org>,
        " Spread Spectrum "<ss@tapr.org>,
        "BBS SIG list   mailing"<bbssig@tapr.org>,
        " tacgps "<tacgps@tapr.org>
Message-ID: <v04003a09b19601d3362e@[207.8.125.50]>
Rendered body type: text/plain

The TAPR office will be closed for at least two weeks beginning Monday.

Dorothy Jones, KA5DWR, Office Manager underwent surgery this morning and
will be in the hospital the rest of this week recovering. Good news is
that the surgery went well and we are waiting for Dorothy now to get out of
post-op.

E-Mails, Faxes, and Calls to the office will still be accepted, but expect
delays in getting kits and publications shipped until things get back to
normal at the Jone's household. I'll be heading to Denton to help at the
office while Dorothy is recovering, but things will be slowed down until
Dorothy is feeling better.

You can send flowers or greetings to:
        Denton Community Hospital
        207 North Bonnie Brae
        Denton TX 76201
        Telephone: 940 898-7000
        Room 139, Dorothy T Jones

I will be taking e-mail up to her if you want to send something by e-mail.

Greg Jones, WD5IVD
Pres. TAPR

Greg Jones, WD5IVD   Austin, Texas
wd5ivd@tapr.org   http://www.tapr.org/~wd5ivd

Real World, The n.:
        5. Anywhere outside a university. "Poor fellow, he's left UT
Austin and gone into the real world."

29. 1998 ARRL and TAPR Digital Communications Conference

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Sun, 31 May 1998 14:10:14 -0500
To: "HF SIG list   mailing"<hfsig@tapr.org>, " Spread Spectrum "<ss@tapr.org>,
        " TAPR/AMSAT DSP "<dsp@tapr.org>,
        "APRS SIG list   mailing"<aprssig@tapr.org>,
        "NETSIG list   mailing"<netsig@tapr.org>,
        "BBS SIG list   mailing"<bbssig@tapr.org>,
        " tacgps "<tacgps@tapr.org>, aprsnews@tapr.org, mic-e@tapr.org,
        TAPR Regional Freq <regional_freq@tapr.org>
Message-ID: <v04003a01b1975b5f40bd@[207.8.125.50]>
Rendered body type: text/plain

1998 ARRL and TAPR Digital Communications Conference
September 25-27, 1998
Chicago, Illinois (minutes from O'Hare Airport)

Web: http://www.tapr.org/dcc

It's that time again! Mark your calendar and think about what to publish for the upcoming 17th Annual ARRL and TAPR Digital Communications Conference. This is the third year in which the ARRL Digital Communications Conference and TAPR Annual General Meeting are joined into one conference!

The ARRL and TAPR Digital Communications Conference is an international forum for radio amateurs in digital communications, networking, and related technologies, who meet, publish their work, and present new ideas and techniques for discussion. Presenters and attendees have the opportunity to exchange ideas and learn about recent hardware and software advances, theories, experimental results, and practical applications. The Digital Communications Conference is not just for the digital expert, but for digitally-oriented amateurs at all levels of expertise.

The 1998 ARRL and TAPR Digital Communications Conference will be held September 25-27, 1998, in Chicago, Illinois. This year's conference location is the Holiday Inn Rolling Meadows, just minutes from O'Hare Airport.

Not only is the Digital Communications Conference technically stimulating, it is a weekend of fun for all who have more than a casual interest in any of the ham digital communications modes. This includes APRS operators, digital networks, software writers, modem designers, and digital satellite enthusiasts. The ARRL and TAPR Digital Communications Conference is for all levels of digital operators -- a must conference to attend to become active on a national level. Now, more than ever, amateur radio needs this great meeting of the minds because it is important that we demonstrate a continued need for the frequency allocations we have by pushing forward and documenting our achievements. The ARRL and TAPR Digital Communications Conference is one of the few ways to record our accomplishments and challenge each other to do more.

A Conference for the Beginner, Too

The conference is not just for the digital expert. This year's conference again provides multiple session tracks for beginning, intermediate, and advanced presentations on selected topics in digital communications. Topics will include APRS, satellite communications, TCP/IP, digital radio, Spread Spectrum, and many others. Come to the conference and hear these topics presented by the experts! Don't miss this opportunity to listen and talk to others in these areas.

Symposia and Seminars

In addition to the presentation of papers on Saturday, three symposia/seminars will be held. These sessions are provided to allow those with additional time and interest to make the most of the Digital Communications Conference. For those who may have interest in just one symposium or seminar, registration for the conference is not required to attend these activities. This allows maximum flexibility for those who may want to participate during the Digital Communications Conference, but do not have an entire weekend to devote to the event.

The Second APRS National Symposium will be held on Friday and will be moderated by Steve Dimse, K4HG (the developer of javAPRS). It will likely include many APRS software authors, such as Bob Bruninga, WB4APR (the father of APRS), Keith Sproul, WU2Z, Mark Sproul, KB2ICI (the developers of MacAPRS and WinAPRS), Brent Hildebrand, KH2Z (the developer of APRSPLUS), and other nationally-known APRS leaders. Join this group for the afternoon and evening for in-depth discussions and presentations on the current and future status of APRS. This is a unique opportunity to gain insight into this fast-growing digital aspect of amateur operations that combines computers, packet radio, and GPS (Global Positioning System).

Starting late Friday afternoon, a half-day seminar titled "Infrastructureless Technologies in Amateur Radio," presented by Don Lemke, WB9MJN, will cover the pros and cons of DDMA (directivity division multiple access, a proposed infrastructureless technigue) versus hubbed (cellular style) and other infrastructure dependant techniques, the basic problem of

megabaud rate communications into the ham QTH, and some techniques for channel-area acquisition including full duplex/asynchronous versus simplex/synchronous radio/acquisition concepts. This seminar will start late enough on Friday afternoon to allow those flying that day to attend the seminar.

The Sunday morning seminar will be a combination of two different talks that will make one excellent seminar. The first presentation, by Lyle Johnson, WA7GXD, will focus on the upcoming RUDAK digital communications system scheduled to fly on the AMSAT Phase 3D satellite in the near future. Lyle will discuss features, systems design, integration, and experiments. The goal of the presentation is to zero in on the communications capabilities of RUDAK, the performance to be expected from the 153k6 PSK system, the limitations and capacities of the DSP modems, user requirements for receiving RUDAK, requirements for transmitting to RUDAK, and how amateurs can work to make ground operations a reality once the satellite is in operation. Information on how to submit a proposal for experimental time on RUDAK will be presented. The second presentation will be by Tim Shepard, KD1KY, regarding "Packet Radio Networks with Millions or Billions of Stations." This talk will present concepts and materials that make you think about different possibilities of networking in the future. Both Lyle's and Tim's presentations focus on different aspects of future digital communications possibilities.

Third Annual ARRL and TAPR DCC Student Papers Award

ARRL and TAPR especially welcome papers from full-time students to compete for the third annual student papers award. Two $500 travel awards may be given, one in each of the following categories: a) best technical/theory-oriented paper by a student, and b) best educational or community-oriented application paper by a student. The paper should relate directly to a wireless digital communication topic (see the guidelines for more information). Papers co-authored by educators or telecommunications professionals are also eligible for this award as long as a student is the first author. The deadline for receipt of finished student paper manuscript is June 20, 1998. Please note that this deadline is different from the general conference paper submission date. For full details and paper guidelines contact TAPR or browse to http://www.tapr.org/dcc.

Call for Conference Proceeding Papers

Anyone interested in digital communications is invited to submit a paper for publication in the Conference Proceedings. Presentation at the conference is not required for publication. The primary purpose of the conference is to communicate ideas and techniques regarding digital communications. Papers written in an informal style are welcome, as well as those written to academic standards. If you know of someone who is doing great things with digital communications, be sure to tell them about this! Papers are due by August 15, 1998, and should be submitted to Maty Weinberg, ARRL, 225 Main Street, Newington, CT 06111, or via e-mail to lweinberg@arrl.org. The paper submission guidelines are available on-line (http://www.tapr.org/dcc).

Local/International Co-Host

The 1998 ARRL and TAPR Digital Communications Conference will be co-hosted by Chicago Amateur Packet Radio Association (CAPRA). CAPRA is one of the oldest packet radio groups in the US. It focuses on education about amateur packet radio, coordination of joint efforts of hams interested in packet radio, and technical and equipment support for a sound packet network in the Chicago area. They have had high-speed packet networks operational since the mid-1980s. Visit the CAPRA Web site at http://www.imaxx.net/capra for more information.

PRUG (Packet Radio User Group of Japan) will be the first International co-host of the ARRL and TAPR Digital Communications Conference. The conference is proud to welcome PRUG as part of the conference. PRUG will be hosting an informal social Friday evening after the seminar and symposium are completed. Visit http://www.prug.or.jp for more information about the organization.

What can you expect during the 1998 ARRL and TAPR Digital Communications Conference?

* A full day of papers, breakout sessions, and selected topics on Saturday for the beginner to advanced amateur digital enthusiast.

* Three seminars/symposia
  Friday (1 PM)
  - Second Annual APRS National Symposium,
    moderated by Steve Dimse, K4HG

  Friday (3 PM)
  - Infrastructure Technologies in Amateur Radio
    conducted by Don Lemke, WB9MJN

  Sunday (8:30 AM)
  - RUDAK Future Operations and Experimentations
    by Lyle Johnson, WA7GXD
  - Packet Radio Networks with Millions or Billions of Stations
    by Tim Shepard, KD1KY

* Third Annual Student Paper Awards.

* TAPR Membership Meeting

* A banquet with special guest speaker Steve Roberts, N4RVE, of Nomadic Research Labs. Creator of the Winnebiko and Behemoth high-tech recumbent bicycles. His current project is focused on Microship technology (http://www.microship.com/).

* SIGs (Special Interest Groups) get-togethers on Saturday, following the banquet

* Informal get-togethers throughout the weekend

* A facility that is perfect for this type of conference

* Informal engineering discussions/demonstration areas

* Motorola Museum located in the area.

* An event at which the most important new developments in amateur digital communications are announced

* Digital 'movers and shakers' from all over the world in attendance

---

Conclusion

If you have attended a Digital Communications Conference in the past, remember how much fun it was discussing the latest developments into the wee hours! If you have never been to a Digital Communications Conference, then make your plans now to attend and find out how much fun they can be.

There are few activities where your participation can be so much fun and so important! You will be able to get together with colleagues from all over the world and bring each other up to date on your latest work. Experience all this and more for an unforgettable weekend of ham radio and digital communications. Make your travel and lodging arrangements now. We hope to see you at the ARRL and TAPR Digital Communications Conference, September 25-27!

Full information on the conference and hotel information can be obtained by contacting Tucson Amateur Packet Radio, 8987-309 E. Tanque Verde Road #337, Tucson, AZ 85749-9399, phone 940-383-0000, fax 940-566-2544, e-mail tapr@tapr.org, browse www.tapr.org/dcc.

Sincerely,

Steve Ford, WB8IMY, ARRL Conference Co-Chair
Greg Jones, WD5IVD, TAPR Conference Co-Chair
Steve Stroh, N8GNJ, Conference Manager
Mark J. Thompson, WB9QZB, Local Host Liaison
Carl Bergstedt, K9VXW, Local Host Liaison

Note: If you need conference handouts or flyers for ham club meetings, contact TAPR to get what you need!

Hotel Information

Conference presentations, meetings, and seminars will be held at the Holiday Inn Rolling Meadows. Special DCC rooms rates are $85 per night. When making reservations with the hotel, be sure to indicate you are attending the ARRL and TAPR DCC in order to get the discount. It is highly recommended that you book your room prior to arriving. A block of 75 rooms is reserved until August 31, 1998. After the 75 rooms are booked, rooms will only be available in nearby hotels, so be sure to book your rooms early! The hotel provides transportation to and from O'Hare Airport. Please contact the hotel to arrange specific transportation needs.

Holiday Inn Rolling Meadows (conference hotel)
3405 Algonquin Rd
Rolling Meadows, IL 60008
Phone 847-259-5000, fax 847-259-0597

Registration

Contact the TAPR office by phone, fax, or e-mail (numbers and addresses are above) to register or for additional meeting information. MasterCard and VISA are accepted.

- Preregistration (before Sept 1): $42.00 *

- Late registration or at door: $47.00 *

* - Conference registration includes Conference Proceedings, Saturday sessions/meetings, and lunch.

- Saturday Evening Dinner (limited space): $20.00 **

** - Includes dinner, speaker, prize drawing

Seminar/symposia

Note: You do not have to be registered for the conference to attend one or more of the seminars/symposia.

- Friday, 1-8 PM, Second Annual APRS National Symposium (full day)

Registration: $25***

- Friday 3-7 PM, Infrastructure Technologies in Amateur Radio, conducted by Don Lemke, WB9MJN

Registration: $15***

Sunday 8:30-11 AM and 11:30 AM-2 PM, RUDAK Future Operations and Experimentations by Lyle Johnson, WA7GXD, and Packet Radio Networks with Millions or Billions of Stations by Tim Shepard, KD1KY

Registration $20***

*** Cost of seminar or symposium includes the cost of handout materials.

Contact TAPR to register for the DCC:

Tucson Amateur Packet Radio
8987-309 E. Tanque Verde Road #337
Tucson, AZ 85749-9399.
phone 940-383-0000, fax 940-566-2544
e-mail tapr@tapr.org, browse http://www.tapr.org

June 1998

Original source file: hfsig.9806
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9806

1. SERIOUS!!!! Hoax

```
From: Robert Glassey <robglassey@geocities.com>
Date: Tue, 02 Jun 1998 12:21:57 -0700
To: Paulus <paulstra@xs4all.nl>, hfsig@tapr.org, bitbucket@qth.net
Cc: Peter Martinez <peter.martinez@btinternet.com>,
        Pesi Sorab <Pesi_Sorab@compuserve.com>,
        Manfred Salzwedel <masa@rack.improvers.fi>,
        Jacques Belfroid <pa3gww@open.net>, Jaap Both <jaapboth@hetnet.nl>,
        Izak Davidse <pa0ihd@pi.net>, Hans Chmielus <JCHMIELUS@aol.com>,
        Ger Rys <rys@nedernet.nl>, Ger Metselaar <ger@mecom.nl>,
        Geoff Bagley <geoff@gcbagley.demon.co.uk>, G3JYF <G3JYF@aol.com>,
        G3IRM <plumb@clara.net>, dl1zam <dl1zam@t-online.de>,
        dennis edey <dennis_edey@interworx.com.au>,
        Cees Slager <cees@mecom.nl>, Brian Beezley <k6sti@n2.net>,
        Bill de Carle <bill@ietc.ca>, Andrew Lovell <lovelinf@algonet.se>,
        Andrew Lovell <andrew@lovell-info.se>,
        Adrew Senior <ASenior383@aol.com>
Message-ID: <357450D5.3DA4@geocities.com>
Rendered body type: text/plain
```

Hi everyone,

This is known as the "Win A Holiday" Hoax, information on this hoax can
be found at URL:

http://ciac.llnl.gov/ciac/CIACHoaxes.html

This is a very handy URL for administrators and anyone else who receives
a Email like this.

> Subject: E-Mail Message Awareness
>
> This alert came from our system administrator....
>
> Be aware!!!!!!
>
> If you receive an e-mail entitled "WIN A HOLIDAY"
> DO NOT open it. It will erase everything on your hard drive.
>
> Forward this letter out to as many people as you can.
> This is a new, very malicious virus and not many people know
> about it.
>
> This information was announced yesterday morning from Microsoft;
> please share it with everyone that might access the internet.
> Once again, pass this along to everyone in your address book so
> that this may be stopped.
>

2. Dorothy Update

```
From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Mon, 8 Jun 1998 10:37:20 -0500
To: "HF SIG  list   mailing"<hfsig@tapr.org>, " Spread Spectrum "<ss@tapr.org>,
        " TAPR/AMSAT DSP "<dsp@tapr.org>,
        "APRS SIG  list   mailing"<aprssig@tapr.org>,
        "NETSIG  list   mailing"<netsig@tapr.org>,
        "BBS SIG  list   mailing"<bbssig@tapr.org>,
        " tacgps "<tacgps@tapr.org>, aprsnews@tapr.org, mic-e@tapr.org,
        TAPR Regional Freq <regional_freq@tapr.org>
Message-ID: <v04003a0cb1a1b453317a@[207.43.172.66]>
Rendered body type: text/plain
```

For those who want to know.

Dorothy was released from the hospital last Friday afternoon and has been
resting at the house (TAPR Office ;-) since then. The surgery went very
well and there appears to be no complications, except that she doesn't like
the really long scar on her belly.

We (Bill Jones and I) have told Dorothy she cannot step one foot into the
TAPR office until next week sometime. Thus, expect kits and orders to
start getting out the door in another 7-10 days. There is a bit of a pile
being generated by this down time, so it will probably be a week or so

after we open back up that we get caught up on everything.

Dorothy wants to thank everyone that thought of her during this ordeal and she looks forward to getting back into the swing of things.

Cheers - Greg Jones, WD5IVD

Greg Jones, WD5IVD Austin, Texas
wd5ivd@tapr.org http://www.tapr.org/~wd5ivd

Real World, The n.:
5. Anywhere outside a university. "Poor fellow, he's left UT Austin and gone into the real world."

3. EVM56002's for sale?

From: Robert Glassey <robglassey@geocities.com>
Date: Wed, 10 Jun 1998 00:37:17 -0700
To: hfsig@tapr.org
Message-ID: <357E37AC.7F22@geocities.com>
Rendered body type: text/plain

Hi all,

A while back I reacll TAPR put together a group order for the Motorola EVM56002 for around $85 rather than the $150 they normally go for. I'm considering getting one of these units and wonder if there is enough interest in putting to gether another group order, TAPR willing of course.

I expect no extra units were purchased for stock, but maybe there are some of these units out there gathering dust and looking for a potential second hand buyer at a good price?

Cheers,

Rob, ZL2AKM

4. Re: [HFSIG:2704] EVM56002's for sale?

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Tue, 9 Jun 1998 10:38:14 -0500
To: hfsig@tapr.org
Message-ID: <v04003a04b1a3072798bb@[207.8.125.50]>
Rendered body type: text/plain

The group purchase was a one time special deal we worked with Motorola. With all the changes within the group at Motorola we worked with, I doubt we will see it happen again.

The $150 price available from several distributors (Wyle, Marshall, etc) is a very good price.

Greg

>Hi all,
>
>A while back I reacll TAPR put together a group order for the Motorola
>EVM56002 for around $85 rather than the $150 they normally go for. I'm
>considering getting one of these units and wonder if there is enough
>interest in putting to gether another group order, TAPR willing of
>course.
>
>I expect no extra units were purchased for stock, but maybe there are
>some of these units out there gathering dust and looking for a potential
>second hand buyer at a good price?
>
>Cheers,
>
>Rob, ZL2AKM

Greg Jones, WD5IVD Austin, Texas
wd5ivd@tapr.org http://www.tapr.org/~wd5ivd

Real World, The n.:
5. Anywhere outside a university. "Poor fellow, he's left UT

Austin and gone into the real world."

5. Re: [HFSIG:2704] EVM56002's for sale?

From: Tim Baggett <tim@acca.nmsu.edu>
Date: Wed, 10 Jun 1998 20:20:39 -0600 (MDT)
To: hfsig@tapr.org
Message-ID: <Pine.BSF.3.96.980610201833.5349A-100000@acca.NMSU.Edu>
Rendered body type: text/plain

On Tue, 9 Jun 1998, Robert Glassey wrote:

> A while back I reacll TAPR put together a group order for the Motorola
> EVM56002 for around $85 rather than the $150 they normally go for. I'm
> considering getting one of these units and wonder if there is enough
> interest in putting to gether another group order, TAPR willing of
> course.

Rob, (and others)

I refer you to Motorola's newest distributor at www.montereytools.com.
They have the 56002EVM listed as $149. You may also check the other
distributors that may be in your own country as well from the Motorola DSP
wed site www.mot.com/SPS/DSP (select distributors from the quick menu at
the bottom). Other distributors include Arrow, and Wiley.

73rds,
Tim

6. VMSK/2 ???

From: "Marcelo Puhl" <mark@plug-in.com.br>
Date: Thu, 11 Jun 1998 21:24:30 -3
To: hfsig@tapr.org
Cc: dsp@tapr.org
Message-ID: <199806120023.VAA03491@bigbox.plug-in.com.br>
Rendered body type: text/plain

Hello,

Anyone researching the VMSK/2 modulation for HF Packet ?

Marcelo Puhl
mark@plug-in.com.br

7. Re: [HFSIG:2707] VMSK/2 ???

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.tu-darmstadt.de>
Date: Fri, 12 Jun 1998 09:27:45 +0200
To: hfsig@tapr.org
Message-ID: <3580D871.D85C6AD@zeus.uet.e-technik.tu-darmstadt.de>
Rendered body type: text/plain

Marcelo Puhl wrote:
>
> Hello,
>
> Anyone researching the VMSK/2 modulation for HF Packet ?

What is VMSK/2? Is it a special type of MSK?

73' Alexander

--
-------------------------------+---------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| technik u. Elektroakustik | Fax : +49-6151-165545
|
| Merckstrasse 25 | EMail: a.kurpiers@uet.tu-darmstadt.de
|
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
-------------------------------+---------------------------------------

8. FFT windowing in an mFSK modem

From: Charles Brain <chbrain@dircon.co.uk>
Date: Sat, 13 Jun 1998 09:51:40 +0100
To: "hfsig@tapr.org" <hfsig@tapr.org>
Message-ID: <35823D9C.904F44C4@dircon.co.uk>
Rendered body type: text/plain

Hello,

What is the best type of window for multipath tolerance in
an mFSK modem that has been implemented using an FFT?

I have tried 3 windows on a recorded signal with about 2ms
multipath.

a) Rectangular window full size of FFT
b) Hanning window full size of FFT
c) Rectangular window from N/4 to 3N/4 all other bins = 0

The modem uses 8ms symbols and the FFT length is the symbol period.

By far the best results appear to come from window c.
window b was the worst.

I am in the process of final tweaking before I release the software
at the beginning of August so any suggestions would be appreciated.

- Charles G4GUO

9. interfacing PK232 to Kenwood TS850 for direct keyed FSK

From: Bill Bennett N7DZ <n7dz@qsl.net>
Date: Sun, 21 Jun 1998 07:05:58 -0700
To: hfsig@tapr.org
Message-ID: <358D1346.574FADEF@qsl.net>
Rendered body type: text/plain

I am trying to drive the TS850 RTTY jack with a PK232 for direct keyed
FSK. I want to try this as an alternative to AFSK into SSB. I am looking
for someone who has done it and can provide guidance on the appropriate
interface and/or comments about the merits of doing it this way.

I don't think the PK232 FSK output is directly compatible with the TS850
RTTY input. TS850 RTTY input requires switching to ground. The PK232 FSK
output is a TI SN7406N with no pullup. I don't have a data book to look
that part up in, but judging from use of SN7406N elsewhere in the PK232,
a pullup resistor to V+, or some other current sourcing load, is
required for this output to work. From the schematic of the TS850, it
looks like the RTTY input is a fairly high impedance switch input, 10k
ohms or so, which I don't believe will source enough current for the
SN7406N to act as a switch.

Does this sound on track at all? Anybody have any advice?

Thanks, 73,

Bill N7DZ

10. Re: [HFSIG:2710] interfacing PK232 to Kenwood TS850 for direct keyed FSK

From: Michael_Joens@Millipore.com
Date: Mon, 22 Jun 1998 14:42:10 -0400
To: hfsig@tapr.org
Message-ID: <8525662B.00639E77.00@MilliGust.Millipore.com>
Rendered body type: text/plain

I don't know anything about the TS850, but I do drive my Icom IC-745 with
direct FSK. All it takes is a very simple homemade interface between RS-232
and the FSK interface of the IC745 to accomplish the level shifting (+/-
12V to TTL). No power needed for the interface. The IC-745 puts out the
Mark signal when the input is open (5-6V) and Space when it is shorted. I
thought of putting an optocoupler in the line to isolate the rig and the
computer, but I found that not to be necessary.

The advantages of true FSK over AFSK into SSB as I see them:

1. Perfectly clean signal (as good as it gets, anyway). No spurious
emissions from insufficient SSB filters, splatter, etc. Fewer conversions
inside the rig, the signal is generated by shifting two frequencies at 9MHz
IF and then converting it to 70MHz and output, vs. AFSK going: digital -
audio - 455kHz - 9Mhz - 70MHz - output.

2. Full power output. No fiddling with audio gain to balance power output with splatter caused by overmodulation. Note: The IC745 supports full FSK output, as do other Icom units I know. Not all brands / models withstand continuous carrier signals, in which case you have to reduce the audio / power gain anyway.

3. The rig can operate in "RTTY" mode for receive. For the IC745, that puts the audio output in the right range, and it applies the right filters. In strong QRM, I can use the 250Hz CW filter to pick out a weak signal and to keep the stronger signal from affecting the AGC.

The only two disadvantages I can see:

1. It uses the serial port, which is also hooked up to the Modem, TNC, Camcorder controller, HT programmer, etc. Prohibits using packet and RTTY at the same time. Can't add another serial port because my PC is full.

2. Modern UARTs don't seem to do 1.5 stop bits. Even though they claim to be 16450 or 16550 compatible, they're not. Those did allow 1.5 stop bits. When I set my PC to 1.5 stop bits, it tries to "fake it" by sending a few bytes with 2 stop bits, then a few with one to get the overall average timing right. This is a somewhat obscure problem that doesn't seem to affect overall performance.

By the way, I'm using the RITTY software by K6STI, and I love it. Does a great job on receive, even if two signals are interlaced. I have used the demo version so far, which times out after 5 minutes. I'm waiting for a friend of mine to compare it to the HamComm software and interface he's using before I buy the license. At $150 it's rather pricey, so I want to make sure it's the right program for me. I don't mind buying the license, but I only want to spend the money once. Since I set it up early May, I've worked about 55 countries with the IC-745 (100 Watts) and a vertical, so I'm pretty happy with it, in spite of the timeout. Anyone else have any experience (good or bad) with the RITTY software?

73,
Michael, KE1JE

Bill Bennett N7DZ <n7dz@qsl.net> on 06/21/98 10:11:52

Please respond to hfsig@tapr.org

To: hfsig@tapr.org
cc: (bcc: Michael Joens/NA/Millipore)
Subject: [HFSIG:2710] interfacing PK232 to Kenwood TS850 for direct keyed FSK

I am trying to drive the TS850 RTTY jack with a PK232 for direct keyed FSK. I want to try this as an alternative to AFSK into SSB. I am looking for someone who has done it and can provide guidance on the appropriate interface and/or comments about the merits of doing it this way.

I don't think the PK232 FSK output is directly compatible with the TS850 RTTY input. TS850 RTTY input requires switching to ground. The PK232 FSK output is a TI SN7406N with no pullup. I don't have a data book to look that part up in, but judging from use of SN7406N elsewhere in the PK232, a pullup resistor to V+, or some other current sourcing load, is required for this output to work. From the schematic of the TS850, it looks like the RTTY input is a fairly high impedance switch input, 10k ohms or so, which I don't believe will source enough current for the SN7406N to act as a switch.

Does this sound on track at all? Anybody have any advice?

Thanks, 73,

Bill N7DZ

11. Re: [HFSIG:2711] Re: interfacing PK232 to Kenwood TS850 for direct

From: Anthony Martin <anthony.martin@gecm.com>
Date: Mon, 22 Jun 1998 20:52:43 +0100
To: hfsig@tapr.org
Message-ID: <358EB60B.4036@gecm.com>
Rendered body type: text/plain

Michael_Joens@millipore.com wrote:
>
> 2. Modern UARTs don't seem to do 1.5 stop bits. Even though they claim to
> be 16450 or 16550 compatible, they're not. Those did allow 1.5 stop bits.
> When I set my PC to 1.5 stop bits, it tries to "fake it" by sending a few
> bytes with 2 stop bits, then a few with one to get the overall average
> timing right. This is a somewhat obscure problem that doesn't seem to
> affect overall performance.
>

While testing modems capable of 5 data 1.5 stop modes we found no
problems with PC UARTS doing this, even on late model Dell pentium
PCs. But I will double-check in future...

If anything not being able to find software or test equipment
supporting 5-bit baudot was the problem. Our program drives the
UARTS directly from DOS.

My DSP software UART does not explicitly check for the presence
of the extra 0.5 of a stop bit. If a terminal device cheated
by using 1 stop bit there could be an unexpectedly high incoming
character rate. Perhaps I should delay searching for the start bit
or add an explicit framing test...

You probably have your PC stuffed full of special multiple-serial
cards with special drivers...

--

A.N.Martin   E-mail anthony.martin@gecm.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK

12. Re: [HFSIG:2712] Re: interfacing PK232 to Kenwood TS850 for direct

From: Michael_Joens@Millipore.com
Date: Mon, 22 Jun 1998 16:59:43 -0400
To: hfsig@tapr.org
Message-ID: <8525662B.00708FDF.00@MilliGust.Millipore.com>
Rendered body type: text/plain

Thanks for the response.

No, my PC is not stuffed with multiple serial port cards, nor am I using
special drivers. RITTY is a DOS program. I'm using the COM2 that's on the
motherboard (and a hardwired external switchbox). It's a Biostar Pentium MB
with a 133MHz processor. I have not been able to confirm that the same
would not happen with a real 16550, even though I have such a board lying
around - simply didn't find it important enough to spend the time on
finding the root cause. So I admit that I jumped to a conclusion here, but
in the scope of my answer I don't think it matters what causes the
phenomenon. I did measure the signal, and confirmed the symptom. I found
that many people use 2 stop bits anyway, and without using diddle, the
length of the stop bit is kind of hard to detect on a scope.

For the discussion of direct FSK I think the stop bit length has little, if
any, relevance. I just mentioned my observation from a purist point of
view. I used to do RTTY on a mechanical typewriter in the 80's, and those
used 1.5 stop bits. German machines normally ran at 50 baud, and we slowed
them down to 45.45baud. I think the US used 1.4 stop bits on the mechanical
machines (or was it the other way around?).

Data books for the 16450 / 16550 that I've seen only show baud rates down
to 50 baud, but of course it's easy to figure out the ratio to get 45 baud
(actually, 22ms bit length). There is a table for configuring byte lenghts
and stop bits, and 1.5 stop bits are only valid for 5 data bits. The
"do-it-all" UART/parallel/floppy chip in my specific motherboard may be
taking an undocumented shortcut, figuring that most applications are now
using 8 bit ASCII. I heard, independently, that one manufacturer's port
chip had a quirk in the implementation of the floppy driver that did not
allow tape drives, so I guess there are, in fact, different levels of
"compatibility".

I'd be interested to see how this one shakes out: hardware or software.

73,
Michael, KE1JE

Anthony Martin <anthony.martin@gecm.com> on 06/22/98 15:59:04

Please respond to hfsig@tapr.org

To: hfsig@tapr.org
cc: (bcc: Michael Joens/NA/Millipore)
Subject: [HFSIG:2712] Re: interfacing PK232 to Kenwood TS850 for direct

Michael_Joens@millipore.com wrote:
>
> 2. Modern UARTs don't seem to do 1.5 stop bits. Even though they claim to
> be 16450 or 16550 compatible, they're not. Those did allow 1.5 stop bits.
> When I set my PC to 1.5 stop bits, it tries to "fake it" by sending a few
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> timing right. This is a somewhat obscure problem that doesn't seem to
> affect overall performance.
>

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PCs. But I will double-check in future...

If anything not being able to find software or test equipment
supporting 5-bit baudot was the problem. Our program drives the
UARTS directly from DOS.

My DSP software UART does not explicitly check for the presence
of the extra 0.5 of a stop bit. If a terminal device cheated
by using 1 stop bit there could be an unexpectedly high incoming
character rate. Perhaps I should delay searching for the start bit
or add an explicit framing test...

You probably have your PC stuffed full of special multiple-serial
cards with special drivers...

--
_____

A.N.Martin E-mail anthony.martin@gecm.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
_____

July 1998

Original source file: hfsig.9807
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9807

1. URGENT NEED Pin layout for Harris RF-5254C

From: "Danny Kohn" <danny.kohn@systematik.se>
Date: Thu, 2 Jul 1998 21:27:50 +0200
To: <hfsig@tapr.org>
Message-ID: <000401bda5ef$8064d320$01c22ac3@pc1.systematik.se>
Rendered body type: text/plain

Hi everyone.
We are going on a fieldday tomorrow and we just received two Harris RF-5254C modems that we would like to test run on the fieldday but we do not have the pin layout on the 15 pin connector.

If you can get us the pin configuration we would be very greatful if you
please mail it to me at danny.kohn@systematik.se.

/73 de (_/)))
Danny Kohn Tel: +46 708 140 300 =('.')=// Miau!
Chairman of Täby Sändaramatörer SK0MT HAM: SM0NBJ (~~~)/Rrrrui!
Member of MillHill ARC, SK0UX ICQ: 1328817 `w---w´
SK0MT: http://ham.te.hik.se/clubs/sk0mt
SK0UX: http://ham.te.hik.se/clubs/sk0ux

2. PSK31 and MT63

From: "L. J. Shepherd" <ljs@west.net>
Date: Wed, 08 Jul 1998 19:09:48 -0700
To: hfsig@tapr.org
Cc: dsp93dev@tapr.org
Message-ID: <3.0.2.32.19980708190948.006ab9c8@west.net>
Rendered body type: text/plain

Hello Group:

I am up and running with my EVM and EVMRI on PSK31 and MT63 hf keyboard to keyboard modes. Please let me know if any of you would like to schedule a test.....any band except 75 meters. Thanks. L. J. Shepherd
K6OYY TAPR 502

3. The Turkey - what and where is it?

From: murrayg@indcom.co.nz
Date: Thu, 9 Jul 1998 15:12:46 +1200
To: hfsig@tapr.org
Message-ID: <199807090307.WAA25212@tapr.org>
Rendered body type: text/plain

Hi,
Does anyone else hear a strange signal that covers about 6950 to 7050 kHz,
almost 24 hrs/day, that sounds a little like the sound a turkey makes? I have determined that it is a series of tones in seemingly random order, each
one on for 20-30 ms, on different frequencies as close as a few hundred Hz,
but since they are spread over such a wide range, the complete transmission
does not fit in my receiver bandwidth.

The signal transmits in bursts about one second long every 45 seconds, and
seems to fade out for a while at local midday. During the evening here in northern ZL it is about S8-S9.

I would be VERY interested in hearing from anyone else who hears it, preferably with their location, the signal strength and if possible, a bearing. If anyone actually KNOWS what it is and who it is, I'd like to hear the details too. Our local Intruder Watch has been unable to identify it through official channels.

Email me as149@detroit.freenet.org

73,
Murray ZL1BPU

4. TAPR EVM56002 Interface boards and Enclosure Kit

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Wed, 8 Jul 1998 22:32:30 -0500
To: "HF SIG list   mailing"<hfsig@tapr.org>
Message-ID: <v04003a02b1c9ea251790@[207.8.125.50]>
Rendered body type: text/plain

Introducing the TAPR EVM56002 Interface boards and Enclosure Kit
http://www.tapr.org/tapr/html/dsp56002evm.html

TAPR is now taking orders for the first round of EVM56002 Interface with enclosure kits until September 15th.

This kit provides:
        An EVM Interface Board Kit and Enclosure for the Motorola EVM56002
        All parts - including screws and standoffs
        Schematics and Documentation (Assembly and Operations)
        Software

        A Motorola EVM56002 (DSP) Interface board IS NOT INCLUDED

The price is:

       $135.00 US for members of TAPR
       or
       $150.00 US for non-members
       + shipping/handling

TAPR kits can be complex depending on the kitting experience of each builder. We don't think you will have trouble with the interface kit, but it does require some knowledge and experience to successfully go from a kit to a finished, usable unit, depending on the mode of operations. For data radio applications (i.e. 9600 baud FSK), special modifications must be made to your radio for proper operation of the EVM interface. In addition, the interface kit requires that you have a Motorola EVM56002 Interface unit. These can be purchased from several electronic distributors (see web page).

TAPR requires that kit purchasers provide VISA/MC information or checks/money orders with their initial purchases. Money for the initial kit purchase will be deposited on September 15th, 1998 to cover kitting costs. Orders will be taken for the first 200 units. If more than 200 units are ordered, then a second or third batch will be done as soon as additional parts inventory can be purchased and kitted. In this way, the interface kit will be provided in the exact numbers required for the demand. After this initial kit offering, EVM interface kits will be provided in batches as the demand warrants doing kits. The initial batch of kits will be as large as the demand requires, which we hope is large. The more the merrier!

EVM interface kit orders for the initial purchase will be taken until September 15th, 1998. Orders can be mailed to the TAPR address: 8987-309 E. Tanque Verde Rd #337, Tucson, Az, 85749-9399, call (940) 383-0000 (Office Hours: Tue-Fri, 9am-12noon, 3pm-5pm Central Time), or fax (940) 566-2544. If you have questions concerning the unit, check the web page.

The interface kit has been discussed for about the last year, with a group coming forward late in 1997 to complete the design process. PSR Issue #69, Winter 1998, has the full description with photos. Current information on the kit is now available at http://www.tapr.org/tapr/html/dsp56002evm.html

This design of the radio interface for the Motorola DSP56002EVM is based on Johan Forrer's, KC7WW, original design. In February 1997, Douglas Braun, N1OWU, released his software suite for the EVM, based on the Finnish Alef Null groups work, and revised his interface to be more flexible than Johan's. The TAPR interface is a happy medium between Johan's and Doug's design. The goal is to make the interface flexible for past and present code developed for the EVM and to create a flexible programming interface for future programmers.

The development team consists of:

Steve Bible, N7HPR	Project Manager, Prototype builder
Lyle Johnson WA7GXD	PCB design and Prototype builder
Johan Forrer KC7WW	Prototype builder
Tim Baggett, AA5DF	Prototype builder
Douglas Braun, N1OWU	Prototype builder
John Koster, W9DDD	PCB Liaison, Beta Tester
Mark Hammond, KC4EBR	Beta Tester
Bdale Garbee, N3EUA	Beta Tester
Keith Justice, KF7TP	Beta Tester
John Ackerman, N8UR	Beta Tester
Frank Perkins, WB5IPM	Beta Tester

```
       Steven Taft, KA1WX        Beta Tester
       David Borden, K8MMO       Beta Tester
       Louis Cobet, K6MDH        Beta Tester
       Jim Lynch, K4GVO          Beta Tester
       L.J. Shepard              Beta Tester
       Greg Jones, WD5IVD        Assistant Project Manager
       Joe Borovetz, WA5VMS      Part Procurement
       Rich Mulvey, AA2YS        Documentation

-------------------------------------------------------------------------------
Tucson Amateur Packet Radio
8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399
-------------------------------------------------------------------------------
e-mail: tapr@tapr.org       web: <http://www.tapr.org/>
ftp:    ftp.tapr.org
phone:  940-383-0000        fax: 940-565-2544
-------------------------------------------------------------------------------
```

5. For Your Eyes Only

```
From: "Lisa" <89333240@19441.com>
Date: Thu, 09 Jul 98 16:24:03 EST
To: Only.Me@19441.com
Message-ID: <007801bda6a2$0673b140$37ab9dcc@ativg.co>
Rendered body type: text/plain
```

```
<HTML><PRE><BODY BGCOLOR="#000000"><FONT COLOR="#00FFFF" SIZE=3>
Hi...

I would like to inform you of a great sex site on the web. it contains everything you will ever dream
of.

There is also a Live Sex Show from Amsterdam, with sound and a chat box.

Go check it out

http://www.angelfire.com/ak/moonshine81/index.html

Loads of Free Videos and Free Pictures

Love

Lisa


</FONT><FONT  COLOR="#000000" SIZE=3>
```

6. Re: [HFSIG:2718] HFSIG digest 527

```
From: "Marcelo Puhl" <mark@plug-in.com.br>
Date: Fri, 10 Jul 1998 01:10:07 -3
To: hfsig@tapr.org
Message-ID: <199807100408.BAA00398@bigbox.plug-in.com.br>
Rendered body type: text/plain
```

```
> ------------------------------------------------------------------------
>
> Date: Thu, 9 Jul 1998 15:12:46 +1200
> From: murrayg@indcom.co.nz
> To: hfsig@tapr.org
> Subject: The Turkey - what  and where is it?
> Message-ID: <199807090307.WAA25212@tapr.org>
>
> Hi, Does anyone else hear a strange signal that covers about 6950 to 7050 kHz,
> almost 24 hrs/day, that sounds a little like the sound a turkey makes? I have
> determined that it is a series of tones in seemingly random order, each one on
> for 20-30 ms, on different frequencies as close as a few hundred Hz, but since
> they are spread over such a wide range, the complete transmission does not fit
> in my receiver bandwidth.
>
> The signal transmits in bursts about one second long every 45 seconds, and
> seems to fade out for a while at local midday. During the evening here in
> northern ZL it is about S8-S9.
>
> I would be VERY interested in hearing from anyone else who hears it,
> preferably with their location, the signal strength and if possible, a
> bearing. If anyone actually KNOWS what it is and who it is, I'd like to hea=
> r the details too. Our local Intruder Watch has been unable to identify it
> through official channels.
>
> Email me as149@detroit.freenet.org
```

>
> 73,
> Murray ZL1BPU
>
>
> ------------------------------

I guess I heard such a signal on 160 m band, 15 years ago. Very weak.

73,
    Marcelo

7. New mode - frequency domain MOSAIC II

From: murrayg@indcom.co.nz
Date: Mon, 13 Jul 1998 15:42:51 +1200
To: hfsig@tapr.org
Message-ID: <199807130335.WAA11817@tapr.org>
Rendered body type: text/plain

If you would like to try a new but very simple digital mode, I have news
for you!

MOSAIC II is a sequential multi-tone Hellschreiber type text mode that
transmits dot-matrix text, upper and lower case, in the frequency
domain, as a series of beeps at different frequencies. Completely
different, and certainly sub-optimal, but intriguing all the same! Multi-tone
Hell is an entirely new philosophy of transmission and reception.

Typical transmissions are under 200Hz wide, and are extremely immune
to noise and interference. Because the transmission is single tone, it
suits CW only transmitters and should be of interest to LF and QRP
operators.

I have demonstrated sending readable text at 10 pixels/sec IN BETWEEN
the two tones of an RTTY station! Two way contacts on HF are easily
achievable under conditions that would destroy RTTY or AMTOR. Signals
can be tape recorded and there is no concept of, or need for, any
synchronism. Hard to believe, but true!

For transmission, all you need is a DOS PC with a speaker - even an old
XT or a palm-top will do. See me for software.

To receive the transmissions you require Win95, a sound card, and
waterfall plot software capable of 25 FFTs per second with 5Hz
resolution, for example GRAM 4.2.2a, freely available from
www.monumental.com/rshorne/gram.html.

Because the receive display is translated by FFT from the frequency
domain, no signal decoding decisions are made by the electronics - it is
then left to your eye and brain to decide what you see. Since humans
are far better than any machine at text pattern recognition, the results
under very low signal-to-noise conditions are remarkable.

If you are interested in experimenting, I have software which transmits
these bird-like chirps from the PC speaker at up to 50 pixels/sec. It can
be demonstrated without a transmitter - as acoustic or ultrasonic
transmission! I am developing a parallel port version to FSK a transmitter
VXO.

Email me at as149@detroit.freenet.org if you are interested in
experimenting. I'd appreciate your feedback.

Murray ZL1BPU

8. Re: [HFSIG:2720] New mode - frequency domain MOSAIC II

From: LareintzJ@aol.com
Date: Mon, 13 Jul 1998 00:29:44 EDT
To: hfsig@tapr.org
Message-ID: <88f413d6.35a98d39@aol.com>
Rendered body type: text/plain

I am interested in experimenting with the MOSAIC II, please send me a copy of
the program, my E-Mail address is Lareintzj@aol.com.   My mailing address is
as follows:

                      LaReintz A. Johnson, II
                      P.O. Box 5032
                      Capitol Heights, Maryland 20791-5032

9. Config PK-900 TNC to IC-751A for PACTOR

From: "Ron Short" <rshort@whidbey.com>
Date: Mon, 13 Jul 1998 07:09:13 -0700
To: <hfsig@tapr.org>
Message-ID: <000b01bdae67$d13de180$71355ecc@svjdbjti>
Rendered body type: text/plain

Well, I am working on getting my PK-900 TNC hooked up to the IC-751A. It is still a challenge for me and as time permits, I will get this figured out. I an trying to set this up using AFSK through the mic plug using a mic to tnc switch. I could use some help with this. Thanks

Ron
KK7GO

rshort@whidbey.com

10. HAMVIEW V 2.05 available

From: Alberto di Bene <dibene@usa.net>
Date: Tue, 14 Jul 1998 15:59:46 +0200
To: dsp@tapr.org, hfsig@tapr.org
Message-ID: <35AB6452.6E9A6673@usa.net>
Rendered body type: text/plain

This is to announce the availability of HAMVIEW V 2.05 at the site
reported below.

WHAT IS HAMVIEW ?

To say it in a few words, HAMVIEW is a program that:

- gets an audio signal either from a Sound Blaster(TM) compatible card
or
  from a .wav file
- performs a windowed spectral analysis on the incoming signal
- display the signal spectrum on the video screen using a carefully
matched
  color scheme
- performs some enhancement on the signal using signal processing
methods
- plays the filtered signal at the output of the audio card
- and last but not least, all these goodies are done in real time!
(well, to
  be honest with about one second of delay from the input to the output,

  due to the data windows used for signal analysis and processing).

The program was especially developed for radio amateurs interested into
all these communication techniques where signals very close to (or more
often below) the receiver noise floor are used, like EME, VLF (137 kHz),
and so on.
However we believe that Hamview will be of value also for HF and VHF CW
lovers (try it... we never found a point-and-click CW filter on a
commercial receiver!!!), and so on.

HAMVIEW is available to all radioamateurs at the following site :
                http://www.freeyellow.com/members/padan

Any feedback is welcome

Alberto di Bene,   I2PHD
Vittorio De Tomasi,   IK2CZL

11. (no subject)

From: Eduardo Jacob <edu@kender.es>
Date: Tue, 14 Jul 1998 22:59:41 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19980714225941.008134a0@kender.es>
Rendered body type: text/plain

I have been very succesfully trying with Peter Martinez G3PLX a new mode
for HF work that he calls MTHell from MultiTone Hell. It has a certain
similitude with Mosaic-II from Murray ZL1BPU. It's an evolution from
Feldhell as he explains here. It's only available for the EVM56002, but
very soon there will be a Sound Blaster version available.

>>> Begin of quote

In about 1940, a system of text transmission over landlines and radio links

was devised in Germany, which transmitted text by transmitting the characters as they appear on the page, using a raster-scan starting at the bottom-left corner and transmitting vertical columns of on-off signals corresponding to black and white. At the receiver, if a signal was detected, a solenoid operated a scanning inking mechanism to reproduce the character on a slowly moving paper tape. There were a number of variants, but the variant known as Feldhell has attracted the interest of radio-amateurs, firstly because a lot of Feldhell machines were collected by amateurs in post-war European countries, and also because the system of transmission itself has some interesting features. More recent Hell activity has used computers and DSP to generate and receive Hell.

In Feldhell, the column scan rate is 17.5 per second. The transmitter and receiver run at the same speed but there is always a random phase-difference between them, which is overcome by displaying two stripes. Each column consists of a sequence of 14 pixels transmitted one after the other, so the baudrate is 245, and there is often a problem with multipath distortion at this speed. With MT Hell, for reasons connected with the DSP setup, the column-scan rate is chosen at 15.625 per sec, but the pixels of each column are spread-out over 250Hz, each pixel having it's own frequency. There is no random sync error, no multipath distortion, and the spectrum of the signal is much more compact that Feldhell.

In this program the column is constructed with 16 tones spaced 15.625Hz apart, but in the same way that you can treat the Feldhell column just as a waveform spread-out in time, you can think of the MT Hell column as being spread-out in frequency and it isn't important how many tones are used to transmit it or receive it, so long as the overall width is right. The resolution works out equal to the bandwidth divided by the column speed. Like Feldhell, there is no decoding in the receiver so there can be no "errors" in the receiver in the sense normally used for digital modes. The decoding is entirely done by the human eye/brain. The original Feldhell machines printed black if the signal was stronger than a certain threshold and white otherwise, but with this program the intensity of the display is proportional to the signal level, and this makes it much easier to read weak signals.

>>> End of quote

I only can say that this mode is very funny and the sound is really strange... perhaps we could try to use it to communicate with whales ;-)

To help people to have and idea of how MTHell (and FeldHell) look like I join a image of a QSO with Peter, there was severe fading but the QSO is still readable.

People interested in MTHell should contact Peter Martinez G3PLX at Peter.Martinez@btinternet.com but remember that it's only for 56002 EVM.

Regards

Eduardo/EA2BAJ

[Attachment: mthell.gif; image/gif; 14,944 bytes]

12. (no subject)

From: "Wolfgang Barthel" <barthel@imsdd.fhg.de>
Date: Thu, 16 Jul 1998 07:16:59 +0200
To: hfsig@tapr.org
Message-ID: <9807160716.ZM8519@sund12>
Rendered body type: text/plain

subscribe
end

August 1998

Original source file: hfsig.9808
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9808

1. Hamview 2.06

```
From: Alberto di Bene <dibene@usa.net>
Date: Sun, 02 Aug 1998 15:56:50 +0200
To: dsp@tapr.org, hfsig@tapr.org
Message-ID: <35C47021.1E8260D1@usa.net>
Rendered body type: text/plain
```

I have just uploaded V 2.06 of Hamview to the site:
http://www.freeyellow.com/members/padan

The reason for the new version is mainly a bug fix. When the environment variable BLASTER was incorrectly set, the program did hang, instead of correctly emitting an error message.

Enjoy,

Alberto, I2PHD

---

2. 1998 ARRL and TAPR Digital Communications Conference

```
From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Sun, 9 Aug 1998 12:20:14 -0600
To: "TAPR-BB  list   mailing"<tapr-bb@tapr.org>
Cc: "HF SIG  list   mailing"<hfsig@tapr.org>, " Spread Spectrum "<ss@tapr.org>,
        " TAPR/AMSAT DSP "<dsp@tapr.org>,
        "APRS SIG  list   mailing"<aprssig@tapr.org>,
        "NETSIG  list   mailing"<netsig@tapr.org>,
        "BBS SIG  list   mailing"<bbssig@tapr.org>,
        " tacgps "<tacgps@tapr.org>, aprsnews@tapr.org, mic-e@tapr.org,
        TAPR Regional Freq <regional_freq@tapr.org>
Message-ID: <v04011707b1f39762e9cf@[168.191.230.109]>
Rendered body type: text/plain
```

1998 ARRL and TAPR Digital Communications Conference
September 25-27, 1998
Chicago, Illinois (minutes from O'Hare Airport)

Web: http://www.tapr.org/dcc

The 1998 ARRL and TAPR Digital Communications Conference will be held September 25-27, 1998, in Chicago, Illinois. This year's conference location is the Holiday Inn Rolling Meadows, just minutes from O'Hare Airport.

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Call for Conference Proceeding Papers

Anyone interested in digital communications is invited to submit a paper for publication in the Conference Proceedings. Presentation at the conference is not required for publication. The primary purpose of the conference is to communicate ideas and techniques regarding digital communications. Papers written in an informal style are welcome, as well

as those written to academic standards. If you know of someone who is doing great things with digital communications, be sure to tell them about this! Papers are due by August 15, 1998, and should be submitted to Maty Weinberg, ARRL, 225 Main Street, Newington, CT 06111, or via e-mail to lweinberg@arrl.org. The paper submission guidelines are available on-line (http://www.tapr.org/dcc).

Local/International Co-Host

The 1998 ARRL and TAPR Digital Communications Conference will be co-hosted by Chicago Amateur Packet Radio Association (CAPRA). CAPRA is one of the oldest packet radio groups in the US. It focuses on education about amateur packet radio, coordination of joint efforts of hams interested in packet radio, and technical and equipment support for a sound packet network in the Chicago area. They have had high-speed packet networks operational since the mid-1980s. Visit the CAPRA Web site at http://www.imaxx.net/capra for more information.

PRUG (Packet Radio User Group of Japan) will be the first International co-host of the ARRL and TAPR Digital Communications Conference. The conference is proud to welcome PRUG as part of the conference. PRUG will be hosting an informal social Friday evening after the seminar and symposium are completed. Visit http://www.prug.or.jp for more information about the organization.

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        by Tim Shepard, KD1KY

*   Third Annual Student Paper Awards.

*   TAPR Membership Meeting

*   A banquet with special guest speaker Steve Roberts, N4RVE, of Nomadic Research Labs. Creator of the Winnebiko and Behemoth high-tech recumbent bicycles. His current project is focused on Microship technology (http://www.microship.com/).

*   SIGs (Special Interest Groups) get-togethers on Saturday, following the banquet

*   Informal get-togethers throughout the weekend

*   A facility that is perfect for this type of conference

*   Informal engineering discussions/demonstration areas

*   Motorola Museum located in the area.

*   An event at which the most important new developments in amateur digital communications are announced

*   Digital 'movers and shakers' from all over the world in attendance

---

Conclusion

If you have attended a Digital Communications Conference in the past,
remember how much fun it was discussing the latest developments into the
wee hours! If you have never been to a Digital Communications Conference,
then make your plans now to attend and find out how much fun they can be.

There are few activities where your participation can be so much fun and so
important! You will be able to get together with colleagues from all over
the world and bring each other up to date on your latest work. Experience
all this and more for an unforgettable weekend of ham radio and digital
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see you at the ARRL and TAPR Digital Communications Conference, September
25-27!

Full information on the conference and hotel information can be obtained by
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Tucson, AZ 85749-9399, phone 940-383-0000, fax 940-566-2544, e-mail
tapr@tapr.org, browse www.tapr.org/dcc.

Sincerely,

   Steve Ford, WB8IMY, ARRL Conference Co-Chair
   Greg Jones, WD5IVD, TAPR Conference Co-Chair
   Steve Stroh, N8GNJ, Conference Manager
   Mark J. Thompson, WB9QZB, Local Host Liaison
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attending the ARRL and TAPR DCC in order to get the discount. It is highly
recommended that you book your room prior to arriving. A block of 75 rooms
is reserved until August 31, 1998. After the 75 rooms are booked, rooms
will only be available in nearby hotels, so be sure to book your rooms
early! The hotel provides transportation to and from O'Hare Airport.
Please contact the hotel to arrange specific transportation needs.

   Holiday Inn Rolling Meadows (conference hotel)
   3405 Algonquin Rd
   Rolling Meadows, IL 60008
   Phone 847-259-5000, fax 847-259-0597

Tucson Amateur Packet Radio
8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399
----
e-mail: tapr@tapr.org   web: <http://www.tapr.org/>
ftp: ftp.tapr.org
phone: 940-383-0000   fax: 940-565-2544
----

3. Re: [HFSIG:2727] 1998 ARRL and TAPR Digital Communications Conference

From: Agari <jon@agari.org>
Date: Sun, 09 Aug 1998 13:36:44 -0400
To: hfsig@tapr.org
Message-ID: <35CDDE2C.E66170EA@agari.org>
Rendered body type: text/plain

will have to miss this one. oh well looked like the "bomb".

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>   Greg Jones, WD5IVD, TAPR Conference Co-Chair
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>
>   Holiday Inn Rolling Meadows (conference hotel)
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>   Rolling Meadows, IL 60008
>   Phone 847-259-5000, fax 847-259-0597
>
> --

```
> Tucson Amateur Packet Radio
> 8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399
> ------------------------------------------------------------------------
> e-mail: tapr@tapr.org       web: <http://www.tapr.org/>
> ftp:    ftp.tapr.org
> phone:  940-383-0000        fax: 940-565-2544
> ------------------------------------------------------------------------
```

4. [HFSIG:2727] 1998 ARRL and TAPR Digital Communications Confe

```
From: Alberto Di Bene <adibene@it.ibm.com>
Date: Mon, 10 Aug 1998 17:51:57 +0000
To: hfsig@tapr.org
Message-ID: <51200100137548000000002L102*@MHS>
Rendered body type: text/plain
```

```
Status Distribution August 09, 1998 17:40:13

The message regarding "[HFSIG:2727] 1998 ARRL and TAPR Digital Communications Confe" sent on August 09,
1998 17:40:13 was sent by

Status Recipient
          Type                To
          Native Name         hfsig@tapr.org
          Foreign Native Name hfsig@tapr.org\n\n\nINTERNET



Recipients

Status Reporters
          Type                From
          Initials            AD
          Name Domain         NOTES
          Native Name         CN=Alberto Di Bene/OU=Italy/O=IBM@IBMIT
          Foreign Native Name CN=Alberto Di Bene/OU=Italy/O=IBM\nIBMIT\n\n
          Organization        IBM
          Org Unit 1          Italy
          Last Name           Bene
          First Name          Alberto



Status          769
Explanation     Invalid recipient

X.400 Status    769
Explanation     Router: Unable to open mailbox file D14HUBM1/14/H/IBM mail.box: Remote system no longer
responding
```

5. [HFSIG:2728] Re: 1998 ARRL and TAPR Digital Communications C

```
From: Alberto Di Bene <adibene@it.ibm.com>
Date: Mon, 10 Aug 1998 18:22:06 +0000
To: hfsig@tapr.org
Message-ID: <51200100137554680000002L182*@MHS>
Rendered body type: text/plain
```

```
Status Distribution August 09, 1998 18:17:27

The message regarding "[HFSIG:2728] Re: 1998 ARRL and TAPR Digital Communications C" sent on August 09,
1998 18:17:27 was sent by

Status Recipient
          Type                To
          Native Name         hfsig@tapr.org
          Foreign Native Name hfsig@tapr.org\n\n\nINTERNET



Recipients

Status Reporters
          Type                From
          Initials            AD
          Name Domain         NOTES
          Native Name         CN=Alberto Di Bene/OU=Italy/O=IBM@IBMIT
          Foreign Native Name CN=Alberto Di Bene/OU=Italy/O=IBM\nIBMIT\n\n
          Organization        IBM
          Org Unit 1          Italy
          Last Name           Bene
```

First Name Alberto

Status 769
Explanation Invalid recipient

X.400 Status 769
Explanation Router: Unable to open mailbox file D14HUBM1/14/H/IBM mail.box: Remote system no longer responding

6. View Amazing World Record Sex!!!

From: knowit98@hotmail.com
Date: Mon, 10 Aug 1998 21:45:58 -0400 (EDT)
Message-ID: <QQfbxf25868.199808110145@relay3.UU.NET>
Rendered body type: text/plain

Attention!
Warning! Adults Only! Warning! Adults Only!

If you are under 21 years of age, or not interested in sexually explicit material...
please hit your keyboard delete button now
and please excuse the intrusion. To REMOVE your name from our
email list send us email with REMOVE in the subject line.
You need not read any further!

Available NOW for only $9.95! Next 10 Days Only!

WORLD RECORD SEX!

Be There! See It Now On Video! Unbelievable ...But True!

You Won't Believe Your Eyes!!! [As Seen on the Howard Stern Show]

"The World's Biggest Gang Bang"

See sexy Annabel Chong as she sets the world Gang Bang Record
in this fantastic video documentary that chronicles her 24 hour
sexathon with 251 men engaging in sexual intercourse and oral
sex with her! Don't worry, you won't have to stay up 24 hours
to watch it all. We've selected only the most exciting and red
hot scenes for you...all in breathtaking living color with plenty of
extreme close-ups! This video is guaranteed to knock your socks
off and leave you breathless! You've never seen anything like it!
Annabel takes on five men at a time! 90 minutes! Order Today!
Only $9.95 plus $3 shipping and handling [Total $12.95].

"GANG BANG II"
The Record Breaker!!! Starring Jasmin St. Claire!

See Beautiful and Voluptious Jasmin St. Claire shatter Annabel's
gang bang record by taking on 300 men in one 24 hour sex session!
You won't believe your eyes at all the hot firey action that you will
see as the new world record is established before your eyes as
Jasmin takes on five men at a time for sexual intercourse and
oral sex! Your friends will break down your door to see this video!
You'll be the most popular guy in town! The action is truly unreal
and you will see the best of it in living life-like color! Order Today
and see Jasmin break the record! 90 minutes. Only $9.95 plus
$3 shipping and handling [total $12.95].

Also Available...

The Uncensored Authentic Underground...
Pamela Anderson Lee & Tommy Lee
Sex Video Tape!

Everyone is talking about this exciting video! See Pam and
Tommy engaging in sexual intercourse and oral sex in the car,
on the boat and much, much more! A real collectors video! 30 minutes.
Only $9.95 plus $3 shipping and Handling [total $12.95]

"Tonya Harding Wedding Night Sex Video"
Now see the beautiful Ice Skating Shame of the Olympics
Tonya Harding engaging in sexual intercourse and oral
sex on her wedding night with husband Jeff Gillooly!

This "Bad Girl" is Hot! Don't miss this video! 30 minutes.
Only $9.95 plus $3 shipping and handling [total $12.95]

"Traci...I Love You" Starring Traci Lords
Now see the most beautiful and popular porn star in her last
adult video before she hit the big time! It's the blockbuster of
the year...sensual...fiery and exposive! Traci Lords in her most
erotic and controversial film ever! Don't Miss It! 90 minutes.
Only $9.95 plus $3 shipping and handling [total $12.95]

EMAIL SPECIAL!

ORDER ANY FOUR VIDEOS AND GET THE FIFTH ONE FREE!!!

Your order will be shipped via First Class Mail. All Shipments in plain unmarked wrapper.
For Priority Mail - Add $5
For Overnight Express - add $15

You can order by Phone, Fax, Mail or Email.

We accept all Major Credit Cards and checks by phone or fax.
Visa - MasterCard - American Express - Discover

10 Day Money Back Guarantee! We know that you will be pleased with these Videos!

To Email your order - DO NOT HIT REPLY ON YOUR KEYBOARD

Send email to our special email address below:

infoplus467@juno.com

[Note: If you order by email and do not receive an email acknowledgement within 24 hours, please phone
our office at
718-287-3800]

Phone our office 9am to 10 pm [eastern time]

[718] 287-3800 to Order By Phone for FASTEST SERVICE!

 We can accept your credit card or check by phone

Fax Your Order 24 hours per day to [718] 462-5920
You can fax your credit card information or your check

Order by mail by sending $12.95 per video, cash, check, money order or major credit card [Visa,
MasterCard, American Express or Discover] to

TCPS, INC.
4718 18th Ave. Suite 135
Brooklyn, NY 11204

Make Checks & Money Orders Payable to TCPS, Inc.
New York State Residents Please Add 85 cents for Sales Tax per Video!

You must be over 21 years of age to order and give us your date of birth with your order!

The Following Order Form is for Your Convenience!

.....
.....

Please ship me the following video tape[s]!

Qty_____Annabel Chong "World's Biggest Gang Bang"

Qty_____"Gang Bang II" Jasmin St. Claire

Qty_____"Pamela & Tommy Lee Sex Video Tape"

Qty_____ "Tonya Harding Wedding Night Sex Video Tape"

Qty_____"Traci I Love You" Traci Lords

at $9.95 each plus $3.00 for shipping and handling per tape
[$12.95 per video or "SPECIAL $51.80 for ALL FIVE"!

Credit Card #_____Exp Date___

I hereby represent that I am over 21 years of age.
My date of birth is_____

Signature_____

Ship to: Name_____

Address_____

City_____State_____Zip_____

Area Code and Home Phone [   ]_____

Fax # [   ]_____

Email Address_____

To remove your name from our mailing list, send us an email with
remove in the subject line. This is a one time offer and you should not hear from us again!

FOREIGN ORDERS -Add $15us if you desire Air Parcel Post Shipment. We ship all over the world.

By deleting your unwanted E-Mail you waste one keystroke, yet
by throwing away paper mail you waste our planet! SAVE THE
TREES and support internet E-Mail instead of paper mail!

[C] Copyright TCPS 1998

7. remove

From: "Nathan Sykes" <nsykes@bellsouth.net>
Date: Mon, 10 Aug 1998 23:51:06 -0500
To: <hfsig@tapr.org>
Message-ID: <000601bdc4e3$a7d4bfc0$2170d6d1@kd4ghg>
Rendered body type: text/plain

-----Original Message-----
From: knowit98@hotmail.com <knowit98@hotmail.com>
To: hfsig@tapr.org <hfsig@tapr.org>
Date: Monday, August 10, 1998 9:19 PM
Subject: [HFSIG:2731] View Amazing World Record Sex!!!

>Attention!
>Warning! Adults Only!   Warning! Adults
Only!
>
>If you are under 21 years of age, or not interested in sexually explicit
material...
>please hit your keyboard delete button now
>and please excuse the intrusion. To REMOVE your name from our
>email list send us email with REMOVE in the subject line.
>You need not read any further!
>
>
>Available NOW for only $9.95! Next 10 Days Only!
>
>WORLD RECORD SEX!
>
>Be There! See It Now On Video! Unbelievable ...But True!
>
>You Won't Believe Your Eyes!!! [As Seen on the Howard Stern Show]
>
>"The World's Biggest Gang Bang"
>
>See sexy Annabel Chong as she sets the world Gang Bang Record
>in this fantastic video documentary that chronicles her 24 hour
>sexathon with 251 men engaging in sexual intercourse and oral
>sex with her! Don't worry, you won't have to stay up 24 hours
>to watch it all. We've selected only the most exciting and red
>hot scenes for you...all in breathtaking living color with plenty of
>extreme close-ups! This video is guaranteed to knock your socks
>off and leave you breathless! You've never seen anything like it!
>Annabel takes on five men at a time! 90 minutes! Order Today!
>Only $9.95 plus $3 shipping and handling [Total $12.95].
>
>
>"GANG BANG II"

>The Record Breaker!!! Starring Jasmin St. Claire!
>
>See Beautiful and Voluptious Jasmin St. Claire shatter Annabel's
>gang bang record by taking on 300 men in one 24 hour sex session!
>You won't believe your eyes at all the hot firey action that you will
>see as the new world record is established before your eyes as
>Jasmin takes on five men at a time for sexual intercourse and
>oral sex! Your friends will break down your door to see this video!
>You'll be the most popular guy in town! The action is truly unreal
>and you will see the best of it in living life-like color! Order Today
>and see Jasmin break the record! 90 minutes. Only $9.95 plus
>$3 shipping and handling [total $12.95].
>
>Also Available...
>
>The Uncensored Authentic Underground...
>Pamela Anderson Lee & Tommy Lee
>Sex Video Tape!
>
>Everyone is talking about this exciting video! See Pam and
>Tommy engaging in sexual intercourse and oral sex in the car,
>on the boat and much, much more! A real collectors video! 30 minutes.
>Only $9.95 plus $3 shipping and Handling [total $12.95]
>
>"Tonya Harding Wedding Night Sex Video"
>Now see the beautiful Ice Skating Shame of the Olympics
>Tonya Harding engaging in sexual intercourse and oral
>sex on her wedding night with husband Jeff Gillooly!
>This "Bad Girl" is Hot! Don't miss this video! 30 minutes.
>Only $9.95 plus $3 shipping and handling [total $12.95]
>
>"Traci...I Love You" Starring Traci Lords
>Now see the most beautiful and popular porn star in her last
>adult video before she hit the big time! It's the blockbuster of
>the year...sensual...fiery and exposive! Traci Lords in her most
>erotic and controversial film ever! Don't Miss It! 90 minutes.
>Only $9.95 plus $3 shipping and handling [total $12.95]
>
>EMAIL SPECIAL!
>
>ORDER ANY FOUR VIDEOS AND GET THE FIFTH ONE FREE!!!
>
>Your order will be shipped via First Class Mail. All Shipments in plain
unmarked wrapper.
>For Priority Mail - Add $5
>For Overnight Express - add $15
>
>You can order by Phone, Fax, Mail or Email.
>
>We accept all Major Credit Cards and checks by phone or fax.
>Visa - MasterCard - American Express - Discover
>
>10 Day Money Back Guarantee! We know that you will be pleased with these
Videos!
>
>To Email your order - DO NOT HIT REPLY ON YOUR KEYBOARD
>
>Send email to our special email address below:
>
>infoplus467@juno.com
>
>[Note: If you order by email and do not receive an email acknowledgement
within 24 hours, please phone our office at
>718-287-3800]
>
>Phone our office 9am to 10 pm [eastern time]
>
>[718] 287-3800 to Order By Phone for FASTEST SERVICE!
>
> We can accept your credit card or check by phone
>
>Fax Your Order 24 hours per day to [718] 462-5920
>You can fax your credit card information or your check
>
>Order by mail by sending $12.95 per video, cash, check, money order or
major credit card [Visa, MasterCard, American Express or Discover] to
>
>TCPS, INC.
>4718 18th Ave. Suite 135

>Brooklyn, NY 11204
>
>Make Checks & Money Orders Payable to TCPS, Inc.
>New York State Residents Please Add 85 cents for Sales Tax per Video!
>
>You must be over 21 years of age to order and give us your date of birth
with your order!
>
>The Following Order Form is for Your Convenience!
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>
>Please ship me the following video tape[s]!
>
>Qty_____Annabel Chong "World's Biggest Gang Bang"
>
>Qty_____"Gang Bang II" Jasmin St. Claire
>
>Qty_____"Pamela & Tommy Lee Sex Video Tape"
>
>Qty_____ "Tonya Harding Wedding Night Sex Video Tape"
>
>Qty_____"Traci I Love You" Traci Lords
>
>at $9.95 each plus $3.00 for shipping and handling per tape
>[$12.95 per video or "SPECIAL $51.80 for ALL FIVE"!
>
>Credit Card #_____Exp Date___
>
>I hereby represent that I am over 21 years of age.
>My date of birth is_____
>
>Signature_____
>
>Ship to: Name_____
>
>Address_____
>
>City_____State_____Zip_____
>
>Area Code and Home Phone [   ]_____
>
>Fax # [   ]_____
>
>Email Address_____
>
>To remove your name from our mailing list, send us an email with
>remove in the subject line. This is a one time offer and you should not
hear from us again!
>
>FOREIGN ORDERS -Add $15us if you desire Air Parcel Post Shipment. We ship
all over the world.
>
>By deleting your unwanted E-Mail you waste one keystroke, yet
>by throwing away paper mail you waste our planet! SAVE THE
>TREES and support internet E-Mail instead of paper mail!
>
>[C] Copyright TCPS 1998
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8. Re: [HFSIG:2731] Unsubscribe

From: Michael_Joens@Millipore.com
Date: Tue, 11 Aug 1998 08:19:14 -0400
To: hfsig@tapr.org
Message-ID: <8525665D.00438AEB.00@MilliGust.Millipore.com>
Rendered body type: text/plain

This is getting sick. Please take me off the spam mailing list immediately.

9. Re: [HFSIG:2731] View Amazing World Record Sex!!!

From: "jsr" <kc5jgv@bobcats2.oursc.k12.ar.us>
Date: Tue, 11 Aug 1998 07:00:42 -0500
To: <hfsig@tapr.org>
Message-ID: <01bdc51f$bfd04f40$6e5ed3aa@kc5jgv.bobcats2.oursc.k12.ar.us>
Rendered body type: text/plain

Who runs this SIG? Can you not keep trash like this off the list? I wouldn't email anything to these folks! They claim to be on Juno. Anyone know how to contact Juno and inform them of this crap?
73,
Scott
KC5JGV

-----Original Message-----
From: knowit98@hotmail.com <knowit98@hotmail.com>
To: hfsig@tapr.org <hfsig@tapr.org>
Date: Monday, August 10, 1998 4:13 PM
Subject: [HFSIG:2731] View Amazing World Record Sex!!!

>Attention!
>Warning! Adults Only! Warning! Adults Only!
>
>If you are under 21 years of age, or not interested in sexually explicit material...
>please hit your keyboard delete button now
>and please excuse the intrusion. To REMOVE your name from our
>email list send us email with REMOVE in the subject line.
>You need not read any further!
>
>
>Available NOW for only $9.95! Next 10 Days Only!
>
>WORLD RECORD SEX!
>
>Be There! See It Now On Video! Unbelievable ...But True!
>
>You Won't Believe Your Eyes!!! [As Seen on the Howard Stern Show]
>
>"The World's Biggest Gang Bang"
>
>See sexy Annabel Chong as she sets the world Gang Bang Record
>in this fantastic video documentary that chronicles her 24 hour
>sexathon with 251 men engaging in sexual intercourse and oral
>sex with her! Don't worry, you won't have to stay up 24 hours
>to watch it all. We've selected only the most exciting and red
>hot scenes for you...all in breathtaking living color with plenty of
>extreme close-ups! This video is guaranteed to knock your socks
>off and leave you breathless! You've never seen anything like it!
>Annabel takes on five men at a time! 90 minutes! Order Today!
>Only $9.95 plus $3 shipping and handling [Total $12.95].
>
>
>"GANG BANG II"
>The Record Breaker!!! Starring Jasmin St. Claire!
>
>See Beautiful and Voluptious Jasmin St. Claire shatter Annabel's
>gang bang record by taking on 300 men in one 24 hour sex session!
>You won't believe your eyes at all the hot firey action that you will
>see as the new world record is established before your eyes as
>Jasmin takes on five men at a time for sexual intercourse and
>oral sex! Your friends will break down your door to see this video!
>You'll be the most popular guy in town! The action is truly unreal
>and you will see the best of it in living life-like color! Order Today
>and see Jasmin break the record! 90 minutes. Only $9.95 plus
>$3 shipping and handling [total $12.95].
>
>Also Available...
>
>The Uncensored Authentic Underground...
>Pamela Anderson Lee & Tommy Lee
>Sex Video Tape!
>
>Everyone is talking about this exciting video! See Pam and
>Tommy engaging in sexual intercourse and oral sex in the car,
>on the boat and much, much more! A real collectors video! 30 minutes.
>Only $9.95 plus $3 shipping and Handling [total $12.95]

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>"Tonya Harding Wedding Night Sex Video"
>Now see the beautiful Ice Skating Shame of the Olympics
>Tonya Harding engaging in sexual intercourse and oral
>sex on her wedding night with husband Jeff Gillooly!
>This "Bad Girl" is Hot! Don't miss this video! 30 minutes.
>Only $9.95 plus $3 shipping and handling [total $12.95]
>
>"Traci...I Love You" Starring Traci Lords
>Now see the most beautiful and popular porn star in her last
>adult video before she hit the big time! It's the blockbuster of
>the year...sensual...fiery and exposive! Traci Lords in her most
>erotic and controversial film ever! Don't Miss It! 90 minutes.
>Only $9.95 plus $3 shipping and handling [total $12.95]
>
>EMAIL SPECIAL!
>
>ORDER ANY FOUR VIDEOS AND GET THE FIFTH ONE FREE!!!
>
>Your order will be shipped via First Class Mail. All Shipments in plain
unmarked wrapper.
>For Priority Mail - Add $5
>For Overnight Express - add $15
>
>You can order by Phone, Fax, Mail or Email.
>
>We accept all Major Credit Cards and checks by phone or fax.
>Visa - MasterCard - American Express - Discover
>
>10 Day Money Back Guarantee! We know that you will be pleased with these
Videos!
>
>To Email your order - DO NOT HIT REPLY ON YOUR KEYBOARD
>
>Send email to our special email address below:
>
>infoplus467@juno.com
>
>[Note: If you order by email and do not receive an email acknowledgement
within 24 hours, please phone our office at
>718-287-3800]
>
>Phone our office 9am to 10 pm [eastern time]
>
>[718] 287-3800 to Order By Phone for FASTEST SERVICE!
>
> We can accept your credit card or check by phone
>
>Fax Your Order 24 hours per day to [718] 462-5920
>You can fax your credit card information or your check
>
>Order by mail by sending $12.95 per video, cash, check, money order or
major credit card [Visa, MasterCard, American Express or Discover] to
>
>TCPS, INC.
>4718 18th Ave. Suite 135
>Brooklyn, NY 11204
>
>Make Checks & Money Orders Payable to TCPS, Inc.
>New York State Residents Please Add 85 cents for Sales Tax per Video!
>
>You must be over 21 years of age to order and give us your date of birth
with your order!
>
>The Following Order Form is for Your Convenience!
>
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>
>Please ship me the following video tape[s]!
>
>Qty_____Annabel Chong "World's Biggest Gang Bang"
>
>Qty_____"Gang Bang II" Jasmin St. Claire
>
>Qty_____"Pamela & Tommy Lee Sex Video Tape"

```
>
>Qty_________   "Tonya Harding Wedding Night Sex Video Tape"
>
>Qty__________"Traci I Love You" Traci Lords
>
>at $9.95 each plus $3.00 for shipping and handling per tape
>[$12.95 per video or "SPECIAL $51.80 for ALL FIVE"!
>
>Credit Card #______________________________Exp Date___
>
>I hereby represent that I am over 21 years of age.
>My date of birth is_________________________________
>
>Signature______________________________________________
>
>Ship to:  Name_______________________________________
>
>Address____________________________________________
>
>City________________________State___________Zip________
>
>Area Code and Home Phone  [    ]___________________________
>
>Fax # [     ]______________________________________________
>
>Email Address___________________________________________
>
>To remove your name from our mailing list, send us an email with
>remove in the subject line.  This is a one time offer and you should not
hear from us again!
>
>FOREIGN ORDERS -Add $15us if you desire Air Parcel Post Shipment.  We ship
all over the world.
>
>By deleting your unwanted E-Mail you waste one keystroke, yet
>by throwing away paper mail you waste our planet!  SAVE THE
>TREES and support internet E-Mail instead of paper mail!
>
>[C] Copyright TCPS 1998
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10. Re: Unwelcomed E-Mail

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Tue, 11 Aug 1998 08:06:43 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.980811080336.28128G-100000@www.kelly-afb.org>
Rendered body type: text/plain

All,

I'm not the mail reflector manager but know those who do manage it and I can assure you that they are just as unhappy about the sexually oriented E-Mail you received via the list. The list is _NOT_ moderated so that E-Mail is not slowed down in getting out to all subscribers.

Anyone can send an E-Mail to the HF-SIG mail "reflector", as we call it, and "hit" each of us with undesirable mail.

Please give the mail reflector manager(s) a break and don't be too hard on them...used you delete key and enjoy the good technical "stuff" that comes into you E-Mail box via the reflector.

Best Regards,

Walt DuBose/K5YFW

```
    ======================================================================
    |           In a world without fences, who needs Gates?              |
    |--------------------------------------------------------------------|
    |        If I wanted "Windows", I'd knock a hole in my wall.         |
    ======================================================================
    |                                  |                                 |
    |                                  | The Micro$oft operating system  |
```

```
| Walt DuBose - K5YFW               | didn't get as bad as it is      |
| E-Mail k5yfw@www.kelly-afb.org    | overnight.  I has taken over 10 |
| Business Telephone: (210)925-1171 | years of careful, calculated,   |
|     Home Telephone: (210)696-3196 | development.                    |
|                                   |                                 |
=======================================================================
|   Using MS is like eating high class dog food...its still dog food. |
=======================================================================
```

11. Re: [HFSIG:2731] View Amazing World Record Sex!!!

From: kb7qlh@juno.com (Richard L Homer)
Date: Tue, 11 Aug 1998 09:32:56 EDT
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <19980811.061555.4391.0.kb7qlh@juno.com>
Rendered body type: text/plain

REMOVE

Richard L. Homer RR 140   52H   KB7QLH
1576 North Orante Road
Tombstone, Az 85638-6120
kb7qlh@juno.com (H) rhomer@tombstonepl.lib.az.us (W)

_____
You don't need to buy Internet access to use free Internet e-mail.
Get completely free e-mail from Juno at http://www.juno.com
Or call Juno at (800) 654-JUNO [654-5866]

12. Re: Unsubscribe

From: kb7qlh@juno.com (Richard L Homer)
Date: Tue, 11 Aug 1998 09:48:24 EDT
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <19980811.063123.4391.4.kb7qlh@juno.com>
Rendered body type: text/plain

me to. !!!!!!
Richard L. Homer RR 140   52H   KB7QLH
1576 North Orante Road
Tombstone, Az 85638-6120
kb7qlh@juno.com (H) rhomer@tombstonepl.lib.az.us (W)

On Tue, 11 Aug 1998 07:25:30 -0500 (CDT) Michael_Joens@millipore.com
writes:
>This is getting sick. Please take me off the spam mailing list
>immediately.
>
>
>

_____
You don't need to buy Internet access to use free Internet e-mail.
Get completely free e-mail from Juno at http://www.juno.com
Or call Juno at (800) 654-JUNO [654-5866]

13. RE: [HFSIG:2735] Re: Unwelcomed E-Mail

From: "Lanier, Robert" <rlanier@harris.com>
Date: Tue, 11 Aug 1998 09:59:02 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <275399FB18C4D111871300805FBEB72F01215AEF@CORPMX6>
Rendered body type: text/plain

I have to second his opinion! I'm not sure how the sexually-explicit email got
on the reflector in the first place, unless it was a "mass mailing." I have
received email like this before, but I don't subscribe to any sex stuff
whatsoever. I think I would be speaking for most of us when I say we would be
curious to know how this email found it way here.

Tony KE4ATO

A little nonsense now and then is relished by the wisest man - Willy Wonka

> -----Original Message-----
> From: hfsig@tapr.org [SMTP:hfsig@tapr.org]
> Sent: Tuesday, August 11, 1998 8:09 AM
> To: hfsig@tapr.org
> Subject: [HFSIG:2735] Re: Unwelcomed E-Mail

All,

I'm not the mail reflector manager but know those who do manage it and I

can assure you that they are just as unhappy about the sexually oriented

E-Mail you received via the list. The list is _NOT_ moderated so that
E-Mail is not slowed down in getting out to all subscribers.

Anyone can send an E-Mail to the HF-SIG mail "reflector", as we call it,

and "hit" each of us with undesirable mail.

Please give the mail reflector manager(s) a break and don't be too hard
on them...used you delete key and enjoy the good technical "stuff" that
comes into you E-Mail box via the reflector.

Best Regards,

Walt DuBose/K5YFW

```
========================================================================
            |           In a world without fences, who needs Gates?
|
|------------------------------------------------------------------------|
            |        If I wanted "Windows", I'd knock a hole in my wall.
|
========================================================================
            |                                    |
|
            |                                    | The Micro$oft operating system
|
            | Walt DuBose - K5YFW                | didn't get as bad as it is
|
            | E-Mail k5yfw@www.kelly-afb.org     | overnight.  I has taken over
10 |
            | Business Telephone: (210)925-1171  | years of careful, calculated,
|
            |     Home Telephone: (210)696-3196  | development.
|
            |                                    |
|
========================================================================
            |   Using MS is like eating high class dog food...its still dog
food. |
========================================================================
```

14. Re: Unwelcomed E-Mail

From: kb7qlh@juno.com (Richard L Homer)
Date: Tue, 11 Aug 1998 10:44:49 EDT
To: hfsig@tapr.org
Message-ID: <19980811.072752.4391.13.kb7qlh@juno.com>
Rendered body type: text/plain

Perhaps they should read some of the mail that goes through their system.
 And take what ever legal action is necessary for them to control "what
goes through their system"

REMOVE ME PLEASE

end

Richard L. Homer RR 140   52H   KB7QLH
1576 North Orante Road
Tombstone, Az 85638-6120
kb7qlh@juno.com (H) rhomer@tombstonepl.lib.az.us (W)

On Tue, 11 Aug 1998 08:08:39 -0500 (CDT) Walt DuBose - K5YFW
<k5yfw@www.kelly-afb.org> writes:
>All,
>
>I'm not the mail reflector manager but know those who do manage it and
>I
>can assure you that they are just as unhappy about the sexually

```
>oriented
>E-Mail you received via the list.  The list is _NOT_ moderated so that
>
>E-Mail is not slowed down in getting out to all subscribers.
>
>Anyone can send an E-Mail to the HF-SIG mail "reflector", as we call
>it,
>and "hit" each of us with undesirable mail.
>
>Please give the mail reflector manager(s) a break and don't be too
>hard
>on them...used you delete key and enjoy the good technical "stuff"
>that
>comes into you E-Mail box via the reflector.
>
>Best Regards,
>
>Walt DuBose/K5YFW
>
>
>=========================================================================
>    |          In a world without fences, who needs
Gates?
> |
>
>|-------------------------------------------------------------------|
>    |        If I wanted "Windows", I'd knock a hole in my wall.
>   |
>
>=========================================================================
>    |                                   |
>   |
>    |                                   | The Micro$oft operating
>system  |
>    | Walt DuBose - K5YFW               | didn't get as bad as it is
>   |
>    | E-Mail k5yfw@www.kelly-afb.org    | overnight.  I has taken over
>10 |
>    | Business Telephone: (210)925-1171 | years of careful,
>calculated,   |
>    |     Home Telephone: (210)696-3196 | development.
>   |
>    |                                  |

> |
>
>=========================================================================
>    |   Using MS is like eating high class dog food...its still dog
>food. |
>
>=========================================================================
>
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>


_____________________________________________________________________
You don't need to buy Internet access to use free Internet e-mail.
Get completely free e-mail from Juno at http://www.juno.com
Or call Juno at (800) 654-JUNO [654-5866]
```

15. Remove

```
From: Steve Gunsel <gunsel@gte.net>
Date: Tue, 11 Aug 1998 16:28:24 -0400
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19980811162824.0079be00@mail.gte.net>
Rendered body type: text/plain

At 08:56 PM 8/10/98 -0500, you wrote:
>Attention!
>Warning! Adults Only!                    Warning! Adults
                            Only!
>
>If you are under 21 years of age, or not interested in sexually explicit
material...
>please hit your keyboard delete button now
>and please excuse the intrusion. To REMOVE your name from our
>email list send us email with REMOVE in the subject line.
>You need not read any further!
```

>
>
>Available NOW for only $9.95! Next 10 Days Only!
>
>WORLD RECORD SEX!
>
>Be There! See It Now On Video! Unbelievable ...But True!
>
>You Won't Believe Your Eyes!!! [As Seen on the Howard Stern Show]
>
>"The World's Biggest Gang Bang"
>
>See sexy Annabel Chong as she sets the world Gang Bang Record
>in this fantastic video documentary that chronicles her 24 hour
>sexathon with 251 men engaging in sexual intercourse and oral
>sex with her! Don't worry, you won't have to stay up 24 hours
>to watch it all. We've selected only the most exciting and red
>hot scenes for you...all in breathtaking living color with plenty of
>extreme close-ups! This video is guaranteed to knock your socks
>off and leave you breathless! You've never seen anything like it!
>Annabel takes on five men at a time! 90 minutes! Order Today!
>Only $9.95 plus $3 shipping and handling [Total $12.95].
>
>
>"GANG BANG II"
>The Record Breaker!!! Starring Jasmin St. Claire!
>
>See Beautiful and Voluptious Jasmin St. Claire shatter Annabel's
>gang bang record by taking on 300 men in one 24 hour sex session!
>You won't believe your eyes at all the hot firey action that you will
>see as the new world record is established before your eyes as
>Jasmin takes on five men at a time for sexual intercourse and
>oral sex! Your friends will break down your door to see this video!
>You'll be the most popular guy in town! The action is truly unreal
>and you will see the best of it in living life-like color! Order Today
>and see Jasmin break the record! 90 minutes. Only $9.95 plus
>$3 shipping and handling [total $12.95].
>
>Also Available...
>
>The Uncensored Authentic Underground...
>Pamela Anderson Lee & Tommy Lee
>Sex Video Tape!
>
>Everyone is talking about this exciting video! See Pam and
>Tommy engaging in sexual intercourse and oral sex in the car,
>on the boat and much, much more! A real collectors video! 30 minutes.
>Only $9.95 plus $3 shipping and Handling [total $12.95]
>
>"Tonya Harding Wedding Night Sex Video"
>Now see the beautiful Ice Skating Shame of the Olympics
>Tonya Harding engaging in sexual intercourse and oral
>sex on her wedding night with husband Jeff Gillooly!
>This "Bad Girl" is Hot! Don't miss this video! 30 minutes.
>Only $9.95 plus $3 shipping and handling [total $12.95]
>
>"Traci...I Love You" Starring Traci Lords
>Now see the most beautiful and popular porn star in her last
>adult video before she hit the big time! It's the blockbuster of
>the year...sensual...fiery and exposive! Traci Lords in her most
>erotic and controversial film ever! Don't Miss It! 90 minutes.
>Only $9.95 plus $3 shipping and handling [total $12.95]
>
>EMAIL SPECIAL!
>
>ORDER ANY FOUR VIDEOS AND GET THE FIFTH ONE FREE!!!
>
>Your order will be shipped via First Class Mail. All Shipments in plain
unmarked wrapper.
>For Priority Mail - Add $5
>For Overnight Express - add $15
>
>You can order by Phone, Fax, Mail or Email.
>
>We accept all Major Credit Cards and checks by phone or fax.
>Visa - MasterCard - American Express - Discover
>
>10 Day Money Back Guarantee! We know that you will be pleased with these
Videos!

>
>To Email your order - DO NOT HIT REPLY ON YOUR KEYBOARD
>
>Send email to our special email address below:
>
>infoplus467@juno.com
>
>[Note: If you order by email and do not receive an email acknowledgement
within 24 hours, please phone our office at
>718-287-3800]
>
>Phone our office 9am to 10 pm [eastern time]
>
>[718] 287-3800 to Order By Phone for FASTEST SERVICE!
>
> We can accept your credit card or check by phone
>
>Fax Your Order 24 hours per day to [718] 462-5920
>You can fax your credit card information or your check
>
>Order by mail by sending $12.95 per video, cash, check, money order or
major credit card [Visa, MasterCard, American Express or Discover] to
>
>TCPS, INC.
>4718 18th Ave. Suite 135
>Brooklyn, NY 11204
>
>Make Checks & Money Orders Payable to TCPS, Inc.
>New York State Residents Please Add 85 cents for Sales Tax per Video!
>
>You must be over 21 years of age to order and give us your date of birth
with your order!
>
>The Following Order Form is for Your Convenience!
>
>.....
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>
>Please ship me the following video tape[s]!
>
>Qty_____Annabel Chong "World's Biggest Gang Bang"
>
>Qty_____"Gang Bang II" Jasmin St. Claire
>
>Qty_____"Pamela & Tommy Lee Sex Video Tape"
>
>Qty_____   "Tonya Harding Wedding Night Sex Video Tape"
>
>Qty_____"Traci I Love You" Traci Lords
>
>at $9.95 each plus $3.00 for shipping and handling per tape
>[$12.95 per video or "SPECIAL $51.80 for ALL FIVE"!
>
>Credit Card #_____Exp Date___
>
>I hereby represent that I am over 21 years of age.
>My date of birth is_____
>
>Signature_____
>
>Ship to: Name_____
>
>Address_____
>
>City_____State_____Zip_____
>
>Area Code and Home Phone [   ]_____
>
>Fax # [   ]_____
>
>Email Address_____
>
>To remove your name from our mailing list, send us an email with
>remove in the subject line. This is a one time offer and you should not
hear from us again!
>

>FOREIGN ORDERS -Add $15us if you desire Air Parcel Post Shipment. We ship
all over the world.
>
>By deleting your unwanted E-Mail you waste one keystroke, yet
>by throwing away paper mail you waste our planet! SAVE THE
>TREES and support internet E-Mail instead of paper mail!
>
>[C] Copyright TCPS 1998
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16. Re: [HFSIG:2732] remove

From: "LSC" <LSC@AMTC.NET>
Date: Tue, 11 Aug 1998 18:17:51 -0400
To: <hfsig@tapr.org>
Message-ID: <199808112214.RAA24544@tapr.org>
Rendered body type: text/plain

> From: Nathan Sykes <nsykes@bellsouth.net>
> To: hfsig@tapr.org
> Subject: [HFSIG:2732] remove
> Date: Tuesday, August 11, 1998 12:57 AM
>
>
> -----Original Message-----
> From: knowit98@hotmail.com <knowit98@hotmail.com>
> To: hfsig@tapr.org <hfsig@tapr.org>
> Date: Monday, August 10, 1998 9:19 PM
> Subject: [HFSIG:2731] View Amazing World Record Sex!!!
>
>
> >Attention!
> >Warning! Adults Only!   Warning! Adults
> Only!
> >
> >If you are under 21 years of age, or not interested in sexually explicit
> material...
> >please hit your keyboard delete button now
> >and please excuse the intrusion. To REMOVE your name from our
> >email list send us email with REMOVE in the subject line.
> >You need not read any further!
> >
> >
> >Available NOW for only $9.95! Next 10 Days Only!
> >
> >WORLD RECORD SEX!
> >
> >Be There! See It Now On Video! Unbelievable ...But True!
> >
> >You Won't Believe Your Eyes!!! [As Seen on the Howard Stern Show]
> >
> >"The World's Biggest Gang Bang"
> >
> >See sexy Annabel Chong as she sets the world Gang Bang Record
> >in this fantastic video documentary that chronicles her 24 hour
> >sexathon with 251 men engaging in sexual intercourse and oral
> >sex with her! Don't worry, you won't have to stay up 24 hours
> >to watch it all. We've selected only the most exciting and red
> >hot scenes for you...all in breathtaking living color with plenty of
> >extreme close-ups! This video is guaranteed to knock your socks
> >off and leave you breathless! You've never seen anything like it!
> >Annabel takes on five men at a time! 90 minutes! Order Today!
> >Only $9.95 plus $3 shipping and handling [Total $12.95].
> >
> >
> >"GANG BANG II"
> >The Record Breaker!!! Starring Jasmin St. Claire!
> >
> >See Beautiful and Voluptious Jasmin St. Claire shatter Annabel's
> >gang bang record by taking on 300 men in one 24 hour sex session!
> >You won't believe your eyes at all the hot firey action that you will
> >see as the new world record is established before your eyes as
> >Jasmin takes on five men at a time for sexual intercourse and

> >oral sex! Your friends will break down your door to see this video!
> >You'll be the most popular guy in town! The action is truly unreal
> >and you will see the best of it in living life-like color! Order Today
> >and see Jasmin break the record! 90 minutes. Only $9.95 plus
> >$3 shipping and handling [total $12.95].
> >
> >Also Available...
> >
> >The Uncensored Authentic Underground...
> >Pamela Anderson Lee & Tommy Lee
> >Sex Video Tape!
> >
> >Everyone is talking about this exciting video! See Pam and
> >Tommy engaging in sexual intercourse and oral sex in the car,
> >on the boat and much, much more! A real collectors video! 30 minutes.
> >Only $9.95 plus $3 shipping and Handling [total $12.95]
> >
> >"Tonya Harding Wedding Night Sex Video"
> >Now see the beautiful Ice Skating Shame of the Olympics
> >Tonya Harding engaging in sexual intercourse and oral
> >sex on her wedding night with husband Jeff Gillooly!
> >This "Bad Girl" is Hot! Don't miss this video! 30 minutes.
> >Only $9.95 plus $3 shipping and handling [total $12.95]
> >
> >"Traci...I Love You" Starring Traci Lords
> >Now see the most beautiful and popular porn star in her last
> >adult video before she hit the big time! It's the blockbuster of
> >the year...sensual...fiery and exposive! Traci Lords in her most
> >erotic and controversial film ever! Don't Miss It! 90 minutes.
> >Only $9.95 plus $3 shipping and handling [total $12.95]
> >
> >EMAIL SPECIAL!
> >
> >ORDER ANY FOUR VIDEOS AND GET THE FIFTH ONE FREE!!!
> >
> >Your order will be shipped via First Class Mail. All Shipments in plain
> unmarked wrapper.
> >For Priority Mail - Add $5
> >For Overnight Express - add $15
> >
> >You can order by Phone, Fax, Mail or Email.
> >
> >We accept all Major Credit Cards and checks by phone or fax.
> >Visa - MasterCard - American Express - Discover
> >
> >10 Day Money Back Guarantee! We know that you will be pleased with
these
> Videos!
> >
> >To Email your order - DO NOT HIT REPLY ON YOUR KEYBOARD
> >
> >Send email to our special email address below:
> >
> >infoplus467@juno.com
> >
> >[Note: If you order by email and do not receive an email acknowledgement
> within 24 hours, please phone our office at
> >718-287-3800]
> >
> >Phone our office 9am to 10 pm [eastern time]
> >
> >[718] 287-3800 to Order By Phone for FASTEST SERVICE!
> >
> > We can accept your credit card or check by phone
> >
> >Fax Your Order 24 hours per day to [718] 462-5920
> >You can fax your credit card information or your check
> >
> >Order by mail by sending $12.95 per video, cash, check, money order or
> major credit card [Visa, MasterCard, American Express or Discover] to
> >
> >TCPS, INC.
> >4718 18th Ave. Suite 135
> >Brooklyn, NY 11204
> >
> >Make Checks & Money Orders Payable to TCPS, Inc.
> >New York State Residents Please Add 85 cents for Sales Tax per Video!
> >
> >You must be over 21 years of age to order and give us your date of birth

> with your order!
> >
> >The Following Order Form is for Your Convenience!
> >
>
>.....

> .....
> >
> >
> >
> >
> >Please ship me the following video tape[s]!
> >
> >Qty_____Annabel Chong "World's Biggest Gang Bang"
> >
> >Qty_____"Gang Bang II" Jasmin St. Claire
> >
> >Qty_____"Pamela & Tommy Lee Sex Video Tape"
> >
> >Qty_____ "Tonya Harding Wedding Night Sex Video Tape"
> >
> >Qty_____"Traci I Love You" Traci Lords
> >
> >at $9.95 each plus $3.00 for shipping and handling per tape
> >[$12.95 per video or "SPECIAL $51.80 for ALL FIVE"!
> >
> >Credit Card #_____Exp Date___
> >
> >I hereby represent that I am over 21 years of age.
> >My date of birth is_____
> >
> >Signature_____
> >
> >Ship to: Name_____
> >
> >Address_____
> >
> >City_____State_____Zip_____
> >
> >Area Code and Home Phone [   ]_____
> >
> >Fax # [   ]_____
> >
> >Email Address_____
> >
> >To remove your name from our mailing list, send us an email with
> >remove in the subject line. This is a one time offer and you should not
> hear from us again!
> >
> >FOREIGN ORDERS -Add $15us if you desire Air Parcel Post Shipment. We
ship
> all over the world.
> >
> >By deleting your unwanted E-Mail you waste one keystroke, yet
> >by throwing away paper mail you waste our planet! SAVE THE
> >TREES and support internet E-Mail instead of paper mail!
> >
> >[C] Copyright TCPS 1998
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17. Ignore the Bad EMail

From: "Victor A. Kean, Jr." <vkean@ds.net>
Date: Tue, 11 Aug 1998 20:32:04 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BDC567.1AC651F0@desk.k1lt.ampr.org>
Rendered body type: text/plain

I find the backlash to the unwanted EMail more offensive than the original unwanted EMail.

Vic, K1LT

18. Re: [HFSIG:2741] Re: remove

From: "LSC" <LSC@AMTC.NET>
Date: Tue, 11 Aug 1998 20:59:38 -0400
To: <hfsig@tapr.org>
Message-ID: <199808120055.TAA12696@tapr.org>
Rendered body type: text/plain

please remove

> From: LSC <LSC@amtc.net>
> To: hfsig@tapr.org
> Subject: [HFSIG:2741] Re: remove
> Date: Tuesday, August 11, 1998 6:15 PM
>
>
>
> ---
> > From: Nathan Sykes <nsykes@bellsouth.net>
> > To: hfsig@tapr.org
> > Subject: [HFSIG:2732] remove
> > Date: Tuesday, August 11, 1998 12:57 AM
> >
> >
> > -----Original Message-----
> > From: knowit98@hotmail.com <knowit98@hotmail.com>
> > To: hfsig@tapr.org <hfsig@tapr.org>
> > Date: Monday, August 10, 1998 9:19 PM
> > Subject: [HFSIG:2731] View Amazing World Record Sex!!!
> >
> >
> > >Attention!
> > >Warning! Adults Only!   Warning! Adults
> > Only!
> > >
> > >If you are under 21 years of age, or not interested in sexually
explicit
> > material...
> > >please hit your keyboard delete button now
> > >and please excuse the intrusion. To REMOVE your name from our
> > >email list send us email with REMOVE in the subject line.
> > >You need not read any further!
> > >
> > >
> > >Available NOW for only $9.95! Next 10 Days Only!
> > >
> > >WORLD RECORD SEX!
> > >
> > >Be There! See It Now On Video! Unbelievable ...But True!
> > >
> > >You Won't Believe Your Eyes!!! [As Seen on the Howard Stern Show]
> > >
> > >"The World's Biggest Gang Bang"
> > >
> > >See sexy Annabel Chong as she sets the world Gang Bang Record
> > >in this fantastic video documentary that chronicles her 24 hour
> > >sexathon with 251 men engaging in sexual intercourse and oral
> > >sex with her! Don't worry, you won't have to stay up 24 hours
> > >to watch it all. We've selected only the most exciting and red
> > >hot scenes for you...all in breathtaking living color with plenty of
> > >extreme close-ups! This video is guaranteed to knock your socks
> > >off and leave you breathless! You've never seen anything like it!
> > >Annabel takes on five men at a time! 90 minutes! Order Today!
> > >Only $9.95 plus $3 shipping and handling [Total $12.95].
> > >
> > >
> > >"GANG BANG II"
> > >The Record Breaker!!! Starring Jasmin St. Claire!
> > >
> > >See Beautiful and Voluptious Jasmin St. Claire shatter Annabel's
> > >gang bang record by taking on 300 men in one 24 hour sex session!
> > >You won't believe your eyes at all the hot firey action that you will
> > >see as the new world record is established before your eyes as
> > >Jasmin takes on five men at a time for sexual intercourse and
> > >oral sex! Your friends will break down your door to see this video!
> > >You'll be the most popular guy in town! The action is truly unreal
> > >and you will see the best of it in living life-like color! Order
Today
> > >and see Jasmin break the record! 90 minutes. Only $9.95 plus

> > >$3 shipping and handling [total $12.95].
> > >
> > >Also Available...
> > >
> > >The Uncensored Authentic Underground...
> > >Pamela Anderson Lee & Tommy Lee
> > >Sex Video Tape!
> > >
> > >Everyone is talking about this exciting video! See Pam and
> > >Tommy engaging in sexual intercourse and oral sex in the car,
> > >on the boat and much, much more! A real collectors video! 30
minutes.
> > >Only $9.95 plus $3 shipping and Handling [total $12.95]
> > >
> > >"Tonya Harding Wedding Night Sex Video"
> > >Now see the beautiful Ice Skating Shame of the Olympics
> > >Tonya Harding engaging in sexual intercourse and oral
> > >sex on her wedding night with husband Jeff Gillooly!
> > >This "Bad Girl" is Hot! Don't miss this video! 30 minutes.
> > >Only $9.95 plus $3 shipping and handling [total $12.95]
> > >
> > >"Traci...I Love You" Starring Traci Lords
> > >Now see the most beautiful and popular porn star in her last
> > >adult video before she hit the big time! It's the blockbuster of
> > >the year...sensual...fiery and exposive! Traci Lords in her most
> > >erotic and controversial film ever! Don't Miss It! 90 minutes.
> > >Only $9.95 plus $3 shipping and handling [total $12.95]
> > >
> > >EMAIL SPECIAL!
> > >
> > >ORDER ANY FOUR VIDEOS AND GET THE FIFTH ONE FREE!!!
> > >
> > >Your order will be shipped via First Class Mail. All Shipments in
plain
> > unmarked wrapper.
> > >For Priority Mail - Add $5
> > >For Overnight Express - add $15
> > >
> > >You can order by Phone, Fax, Mail or Email.
> > >
> > >We accept all Major Credit Cards and checks by phone or fax.
> > >Visa - MasterCard - American Express - Discover
> > >
> > >10 Day Money Back Guarantee! We know that you will be pleased with
> these
> > Videos!
> > >
> > >To Email your order - DO NOT HIT REPLY ON YOUR KEYBOARD
> > >
> > >Send email to our special email address below:
> > >
> > >infoplus467@juno.com
> > >
> > >[Note: If you order by email and do not receive an email
acknowledgement
> > within 24 hours, please phone our office at
> > >718-287-3800]
> > >
> > >Phone our office 9am to 10 pm [eastern time]
> > >
> > >[718] 287-3800 to Order By Phone for FASTEST SERVICE!
> > >
> > > We can accept your credit card or check by phone
> > >
> > >Fax Your Order 24 hours per day to [718] 462-5920
> > >You can fax your credit card information or your check
> > >
> > >Order by mail by sending $12.95 per video, cash, check, money order
or
> > major credit card [Visa, MasterCard, American Express or Discover] to
> > >
> > >TCPS, INC.
> > >4718 18th Ave. Suite 135
> > >Brooklyn, NY 11204
> > >
> > >Make Checks & Money Orders Payable to TCPS, Inc.
> > >New York State Residents Please Add 85 cents for Sales Tax per Video!
> > >
> > >You must be over 21 years of age to order and give us your date of

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birth
> > with your order!
> > >
> > >The Following Order Form is for Your Convenience!
> > >
> >
>
>.....................................................................

>
> > ..............................
> > >
> > >
> > >
> > >
> > >Please ship me the following video tape[s]!
> > >
> > >Qty___________Annabel Chong "World's Biggest Gang Bang"
> > >
> > >Qty___________"Gang Bang II" Jasmin St. Claire
> > >
> > >Qty___________"Pamela & Tommy Lee Sex Video Tape"
> > >
> > >Qty___________   "Tonya Harding Wedding Night Sex Video Tape"
> > >
> > >Qty___________"Traci I Love You" Traci Lords
> > >
> > >at $9.95 each plus $3.00 for shipping and handling per tape
> > >[$12.95 per video or "SPECIAL $51.80 for ALL FIVE"!
> > >
> > >Credit Card #_____________________________Exp Date___
> > >
> > >I hereby represent that I am over 21 years of age.
> > >My date of birth is_____________________________
> > >
> > >Signature_____________________________________________
> > >
> > >Ship to:  Name_____________________________________
> > >
> > >Address_____________________________________________
> > >
> > >City_____________________State___________Zip_________
> > >
> > >Area Code and Home Phone  [    ]_________________________
> > >
> > >Fax # [     ]_____________________________________________
> > >
> > >Email Address_____________________________________________
> > >
> > >To remove your name from our mailing list, send us an email with
> > >remove in the subject line.  This is a one time offer and you should
not
> > hear from us again!
> > >
> > >FOREIGN ORDERS -Add $15us if you desire Air Parcel Post Shipment.  We
> ship
> > all over the world.
> > >
> > >By deleting your unwanted E-Mail you waste one keystroke, yet
> > >by throwing away paper mail you waste our planet!  SAVE THE
> > >TREES and support internet E-Mail instead of paper mail!
> > >
> > >[C] Copyright TCPS 1998
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19. Re: Re: remove

From: Bob Johnson <bobj@atlantic.net>
Date: Tue, 11 Aug 1998 22:50:16 -0400
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19980811225016.00712100@mail.atlantic.net>

Rendered body type: text/plain

At 07:58 PM -0500, you wrote:
>please remove

EVERYONE, please understand a few things:

1) When you follow their instructions to reply with a message that says "REMOVE", you are sending it to the HFSIG list, not to them. That just annoys your fellow list members, and doesn't do anything to stop the mail.

2) Even if you were sending a reply to them, you don't want to. If you reply to their message, that marks you as someone who reads junk email, and they will put you on a "good prospect" list so you will get even more email. IF YOU REPLY TO THEIR MAIL, YOU GET EVEN MORE MAIL.

3) One way that these people get email addresses is to have a computer automatically browse web pages looking for email addresses. The other is to automatically browse Usenet news groups. If your email address is published on a web page or on Usenet, it will probably end up on a junk mail list eventually. That's probably how they got this one.

4) Given that it's very difficult to keep your email address off of junk email lists, the most efficient method of dealing with the stuff is to delete it as soon as you realize what it is. Don't even finish reading it, it's a waste of your time. I know, it's frustrating, but the energy you spend complaining about junk email is better spent writing your congress critter about your pet topic (junk email, for example).

And, while I'm at it, it is generally a bad idea to send anything but plain text attachments to a mailing list. For one thing, there are too many dangerous things that can happen if you open unknown attachments with some email programs: you shouldn't open email attachments (other than .txt files) unless you are absolutely sure you know who sent it, and are absolutely sure you trust the sender to install a program on your computer. Executables (.exe files) are a particularly bad idea. In addition, non-text attachments tend to be much larger than is polite to send on an email list (they take too long to download). To distribute large attachments, I recommend putting them on a WWW or FTP server and distributing a short message describing the file and giving the URL where it can be found.

Sorry for the length of this message. I hope it does more good than harm!

-- Bob Johnson
   WB4JCM

20. Re: Error Condition Re: Notorious SPAM

From: CDrentea@aol.com
Date: Tue, 11 Aug 1998 22:57:11 EDT
To: hfsig@tapr.org
Message-ID: <3018cba2.35d10488@aol.com>
Rendered body type: text/plain

REMOVE MY NAME AT ONCE!

21. Remove me from the list!!!

From: "harens" <harens@netcom.ca>
Date: Wed, 12 Aug 1998 01:00:16 -0400
To: <hfsig@tapr.org>
Message-ID: <001401bdc5ae$1edb1ea0$eeb9fea9@home>
Rendered body type: text/plain

[Empty message body]

22. Re: Junk on HFSIG

From: Anthony Martin <anthony.martin@gecm.com>
Date: Wed, 12 Aug 1998 14:51:29 +0100
To: hfsig@tapr.org
Message-ID: <35D19DE1.4B97@gecm.com>
Rendered body type: text/plain

Is the listserv set up to only allow posting from registered
subscribers?

If it is, can the organisers get the spammers' ID off the list?

If it isn't, could it be enabled as a means of stopping the junk?

Spammers may have picked up the hfsig email address while trawling
the web; the're unlikely to bother actively subscribing. This one
didn't even bother with the usual precaution of removing *.org
addresses from the mailing list.

I've had several copies of this spam already, not via HFSIG. If
the've posted here once, It's likely to recur unless blocked.

If I ever mention HFSIG on usenet I always munge the email address.

Ant
--
_____

A.N.Martin   E-mail anthony.martin@gecm.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
_____

23. Unsuscribe

From: "Jorge L. Alvarez" <jallen2@earthlink.net>
Date: Wed, 12 Aug 1998 09:20:11 -0700
To: hfsig@tapr.org
Message-ID: <35D1C0B8.1017F232@earthlink.net>
Rendered body type: text/plain

Please, Unsuscribe
Please remove my name and addres from the mailing list.
Thank.

> Topics covered in this issue include:
>
>   1) Re: Unsubscribe
>   by kb7qlh@juno.com (Richard L Homer)
>   2) Re: Unwelcomed E-Mail
>   by "Lanier, Robert" <rlanier@harris.com>
>   3) Re: Unwelcomed E-Mail
>   by kb7qlh@juno.com (Richard L Homer)
>   4) Remove
>   by Steve Gunsel <gunsel@gte.net>
>   5) Re: remove
>   by "LSC" <LSC@AMTC.NET>
>   6) Ignore the Bad EMail
>   by "Victor A. Kean, Jr." <vkean@ds.net>
>   7) Re: remove
>   by "LSC" <LSC@AMTC.NET>
>
>
[Attachment: vcard.vcf; text/x-vcard; 336 bytes]

24. HOW TO UNSUBSCRIBE

From: Tom McDermott <mcdermot@aud.alcatel.com>
Date: Wed, 12 Aug 1998 13:09:31 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <01BDC5F2.724F5080.mcdermot@aud.alcatel.com>
Rendered body type: text/plain

I'm not associated with this list, but there seem to be a lot of messages from
those who wish to unsubscribe, but do not know how.

To unsubscribe from this list, you send mail to the list-server, not to the mail
reflector.

Send email to the listserver at

listserv@tapr.org

with the subject line of

unsubscribe hfsig

Below is the help text, which you can retrieve by sending a message to
listserv@tapr.org   with the subject of HELP.

Recognized requests are:

```
help [topic]
------------
Without arguments, this file. Otherwise get specific information on the
selected topic. Topics may also refer to requests. To learn more about this
system issue a 'help listproc' request. To get a listing of all available
topics, generate an error message by sending a bogus request like 'help me'.


set <list> [<option> <arg[s]>]
------------------------------
Without the optional arguments, get a list of all current settings for
the specified list. Otherwise change the option to a new value for that
list. Issue a 'help set' request for more information.


subscribe <list> <your name>
----------------------------
The only way to subscribe to a list.


unsubscribe <list>
signoff <list>
------------------
Two ways of removing yourself from the specified list.


recipients <list>
review <list>
-----------------
Get a listing of all non-concealed people subscribed to the specified list.


information <list>
------------------
Get information about the specified list.


statistics <list> {[subscriber email address(es)] | [-all]}
-----------------------------------------------------------
Get a listing of non-concealed subscribers along with the number of messages
each one of them has sent to the specified list. If the optional email addresses
are given, then statistics will be collected for these users only. For example:
                stat foo user1@domain user2@domain
will generate statistics about these two subscribers. "-all" lists statistics
for all users that have posted on the list (whether currently subscribed ot
not).

run <list> [<password> <cmd [args]>]
------------------------------------
Run the specified command with the optional arguments and receive the output
from stdout and/or stderr. To get a listing of all available commands to run,
omit the arguments, i.e. issue a 'run <list>' request. You have to belong to
the specified list, and must have obtained the password from the list's owner;
the owner's address may be found in the Errors-To: header line of each
delivered message. <list> may be local only.


lists
-----
Get a list of all local mailing lists that are served by this server, as well
as of all known remote lists.


index [archive | path-to-archive] [/password] [-all]
----------------------------------------------------
Get a list of files in the selected archive, or the master archive if no
archive was specified. If an archive is private, you have to provide its
password as well.


get <archive | path-to-archive> <file> [/password] [parts]
----------------------------------------------------------
Get the requested file from the specified archive. Files are usually split in
parts locally, and in such a case you will receive the file in multiple email
messages -- an 'index' request tells you how many parts the file has been split
into, and their sizes; if you need to obtain certain parts, specify them as
optional arguments. If an archive is private, you have to provide its password
as well.
```

```
view <archive | path-to-archive>] [/password] [parts]
-----------------------------------------------------
Same as "get" but in interactive mode justs catenates the file on the
screen.

search <archive | path-to-archive>] [/password] [-all] <pattern>
---------------------------------------------------------------
Search all files of the specified archive (and all of its subarchives if -all
is specified) for lines that match the pattern. The pattern can be an
egrep(1)-style regular expression with support for the following additional
operators: '~' (negation), '|' and '&' (logical OR and AND), '<' '>' (group
regular expressions). The pattern may be enclosed in single or double quotes.
Note: . matches any character including new line.

fax <fax-number> <archive | path-to-archive> <file> [/password] [parts]
-----------------------------------------------------------------------
Same as 'get', but it faxes you the files instead to the specified number.

release
-------
Get information about the current release of this ListProcessor system.


which
-----
Get a listing of local mailing lists to which you have subscribed.

=============================================================================

The requests below are list owner specific. List owners should contact the
system's manager for extensive usage information:


reports <list> <password>
-------------------------
Obtain all reports about the specified local list.


edit <list> <password> <file>
-----------------------------
Obtain the specified file for editing.


put <list> <password> <keyword> [args]
--------------------------------------
Manipulate addresses and change system files.


approve <list> <password> <tag>
-------------------------------
Approve the message identified by the tag number for posting to the
specified moderated list.


discard <list> <password> <tag>
-------------------------------
Discard the message identified by the tag number sent to the specified
moderated list.
=============================================================================

In addition, help is available on the following topic(s):

live
----
Learn how to connect to this ListProcessor over the Internet for live
processing of your requests.

listproc
--------
Learn more about this system and list management software in general.

Tom McDermott
mcdermot@aud.alcatel.com
```

25. Re: [HFSIG:2748] Unsuscribe

From: "Jerome M. Kutche" <n9lya@tima.com>
Date: Wed, 12 Aug 1998 15:54:06 -0500

To: hfsig@tapr.org
Message-ID: <35D200EB.355CAFE8@tima.com>
Rendered body type: text/plain

You have sent your request to the group not the list server please read your
instuctions you were given when you subscribe so you know who to send your
request to.... This attempt only wastes out time/

Jorge L. Alvarez wrote:

> Please, Unsuscribe
> Please remove my name and addres from the mailing list.
> Thank.
>
> > Topics covered in this issue include:
> >
> >   1) Re: Unsubscribe
> >   by kb7qlh@juno.com (Richard L Homer)
> >   2) Re: Unwelcomed E-Mail
> >   by "Lanier, Robert" <rlanier@harris.com>
> >   3) Re: Unwelcomed E-Mail
> >   by kb7qlh@juno.com (Richard L Homer)
> >   4) Remove
> >   by Steve Gunsel <gunsel@gte.net>
> >   5) Re: remove
> >   by "LSC" <LSC@AMTC.NET>
> >   6) Ignore the Bad EMail
> >   by "Victor A. Kean, Jr." <vkean@ds.net>
> >   7) Re: remove
> >   by "LSC" <LSC@AMTC.NET>
> >
> >
>
>   --
>
>   Jorge L. Alvarez <jallen2@earthlink.net>
>   Electronic Engineer
>
>   Jorge L. Alvarez
>   Electronic Engineer   <jallen2@earthlink.net>
>   3428 W. Orange Ave. 3208 Home: 714-484-9954
>   Anaheim   Netscape Conference Address
>   CA   Netscape Conference DLS Server
>   92804
>   Additional Information:
>   Last Name   Alvarez
>   First Name   Jorge L.
>   Version   2.1

26. Any HF data activity?

From: Gary Chatters <gc@universe.digex.net>
Date: Wed, 12 Aug 1998 17:58:56 -0400
To: hfsig@tapr.org
Message-ID: <199808122158.RAA20924@universe.digex.net>
Rendered body type: text/plain

This list (HFSIG) used to be about the most interesting mail list in the net.

I didn't participate myself, but it gave me some incentive to at least
think about improving my HF capability to the point where I could try
some of the more interesting modes. Right now I am stuck with Baudot RTTY,
receive only and not very good at that. (It is kind of fun, though).

But then there was the loss of the list and it has never recovered.
The only list activity has been the recent spam followed by a
flurry of net newbies sending unsubscribe requests for us to read.

Is there anyone on this list actually doing HF data experiments?
Have they all moved to some other list?

Maybe there were only a couple of people doing it anyway and they
lost interest.

73,
Gary

27. PLEASE READ re: SPAM

From: Mike Fahmie <MPFahmie@lbl.gov>
Date: Wed, 12 Aug 1998 15:05:32 -0700

To: hfsig@tapr.org
Message-ID: <199808122207.PAA10471@mh1.lbl.gov>
Rendered body type: text/plain

Many of you are sending REMOVE requests in response to the recent spam attack on the SIG list. This is not necessary because the spam was sent to the LIST SERVER, not to you personally. The spammer DOES NOT HAVE your personal email address (yet).

There have been many REMOVE requests sent erroneously to the LIST, evidently using the email REPLY function, perhaps others have sent their requests to the original sender. Remember, the original sender DOES NOT HAVE your personal address, but he WILL HAVE IT if you respond to him.

-Mike-
WA6ZTY

28. Re: [HFSIG:2748] Unsuscribe

From: David A Fischer <lindburg@earthlink.net>
Date: Wed, 12 Aug 1998 17:26:55 -0500
To: hfsig@tapr.org
Message-ID: <35D216AF.41353C48@mail.earthlink.net>
Rendered body type: text/plain

Jorge L. Alvarez wrote:

> Please, Unsuscribe
> Please remove my name and addres from the mailing list.
> Thank.
>
> > Topics covered in this issue include:
> >
> >   1) Re: Unsubscribe
> >   by kb7qlh@juno.com (Richard L Homer)
> >   2) Re: Unwelcomed E-Mail
> >   by "Lanier, Robert" <rlanier@harris.com>
> >   3) Re: Unwelcomed E-Mail
> >   by kb7qlh@juno.com (Richard L Homer)
> >   4) Remove
> >   by Steve Gunsel <gunsel@gte.net>
> >   5) Re: remove
> >   by "LSC" <LSC@AMTC.NET>
> >   6) Ignore the Bad EMail
> >   by "Victor A. Kean, Jr." <vkean@ds.net>
> >   7) Re: remove
> >   by "LSC" <LSC@AMTC.NET>
> >
> >
>
>   --
>
>   Jorge L. Alvarez <jallen2@earthlink.net>
>   Electronic Engineer
>
>   Jorge L. Alvarez
>   Electronic Engineer   <jallen2@earthlink.net>
>   3428 W. Orange Ave. 3208 Home: 714-484-9954
>   Anaheim   Netscape Conference Address
>   CA   Netscape Conference DLS Server
>   92804
>   Additional Information:
>   Last Name   Alvarez
>   First Name   Jorge L.
>   Version   2.1

29. Radio and Network set up

From: Agari Communication <jon@agari.org>
Date: Wed, 12 Aug 1998 18:42:16 -0400
To: hfsig@tapr.org
Message-ID: <35D21A48.83331C5E@agari.org>
Rendered body type: text/plain

This is a general question about the set up the software with out a radio. I would like to set a workstation on my LAN to be able to decode signals such as faxes. Can I configure a linux workstation to do due this through a sound card and a HF receiver ? ALso can I set up a work station as if it where connected to a TNC without the TNC present, simply to set up the software, or would that be a wasted effort ?

Jonathan Phillips
Network Administation

30. Re: [HFSIG:2733] Re: Unsubscribe

From: David A Fischer <lindburg@earthlink.net>
Date: Wed, 12 Aug 1998 17:25:19 -0500
To: hfsig@tapr.org
Message-ID: <35D2164F.F1F10F08@mail.earthlink.net>
Rendered body type: text/plain

Michael_Joens@millipore.com wrote:

> This is getting sick. Please take me off the spam mailing list immediately.

31. Re: [HFSIG:2754] Radio and Network set up

From: forrerj@peak.org (Johan Forrer)
Date: Wed, 12 Aug 1998 18:00:33 -0800
To: hfsig@tapr.org
Message-ID: <199808130103.SAA13893@peak.org>
Rendered body type: text/plain

Johathan,

I am in the process of bring up something along those lines. At bit
of background ....it is a networked cluster of Linux machines, with
the master node running the display functionality under X-windows
and Motif-based applications. It controls various processes running
on this multiprocessor system. Some of the Linux systems are dedicated
to control and run communications-related tasks. Its really a nice
way extend and use those older computers gathering dust.

You may even be able, for example to do some native DSP work
on an Pentium class node equipped with the appropiate hardware.

Perhaps you may want to look at the work being done using PVM at
www.linux.org and other places.

Trust this is of interest.

73

Johan, KC7WW

>This is a general question about the set up the software with out a
>radio. I would like to set a workstation on my LAN to be able to decode
>signals such as faxes. Can I configure a linux workstation to do due
>this through a sound card and a HF receiver ? ALso can I set up a work
>station as if it where connected to a TNC without the TNC present,
>simply to set up the software, or would that be a wasted effort ?
>
>
>Jonathan Phillips
>Network Administation
>
>
>

32. Human readable digital modes

From: murrayg@indcom.co.nz
Date: Thu, 13 Aug 1998 13:54:47 +1200
To: hfsig@tapr.org
Message-ID: <199808130147.NAA63986@iprolink.co.nz>
Rendered body type: text/plain

Hi,
A new site of interest to Amateurs with a digital bent has just started up.

It is dedicated to human readable visual text modes, of which two are
currently known. Most folk do not even know of their existence!

* Feld - Hell
* Concurrent multi-tone Hell
* Sequential multi-tone Hell

These modes can provide high performance in narrow bandwidth under
very noisy conditions, and benefit greatly from DSP techniques to
provide the fuzzies. Feld-Hell has a good following in Europe - multi-tone

Hell is all new!

To enrich your experience of Amateur Radio, check out the site:

http://members.xoom/ZL1BPU/Contents.html

73,

Murray ZL1BPU

33. Re: SPAM: Don't make it worse by hitting REPLY.....

From: DSCHULTZ@SSSP.HST.NASA.GOV
Date: Thu, 13 Aug 1998 3:48:51 GMT
To: hfsig@tapr.org
Cc: DSCHULTZ@SSSP.HST.NASA.GOV
Message-ID: <980813034851.2bf@SSSP.HST.NASA.GOV>
Rendered body type: text/plain

Mike, WA6ZTY, wrote:

>Many of you are sending REMOVE requests in response to the recent spam
>attack on the SIG list. This is not necessary because the spam was sent to
>the LIST SERVER, not to you personally. The spammer DOES NOT HAVE your
>personal email address (yet).

>There have been many REMOVE requests sent erroneously to the LIST, evidently
>using the email REPLY function, perhaps others have sent their requests to
>the original sender. Remember, the original sender DOES NOT HAVE your
>personal address, but he WILL HAVE IT if you respond to him.

In addition to the above mentioned gaffe, improper use of the REPLY function
ends up attaching a complete copy of the offending SPAM message to your
misdirected REMOVE request. The end result is that instead of receiving
just ONE copy of the x-rated spam, I received almost a dozen copies to
take up space in my mailbox while I was away for the past few days.

WOULD YOU GUYS PLEASE READ THE INSTRUCTIONS FOR YOUR E-MAIL PROGRAM!!!!!
Don't make the SPAM problem worse by mailing extra copies of it back
to the list. I thought TAPR people were smarter than that.....

Dan Schultz N8FGV

34. Re: [HFSIG:2754] Radio and Network set up

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Thu, 13 Aug 1998 09:03:51 +0200
To: hfsig@tapr.org
Message-ID: <35D28FD7.6F96DEE@ife.ee.ethz.ch>
Rendered body type: text/plain

Agari Communication wrote:
>
> This is a general question about the set up the software with out a
> radio. I would like to set a workstation on my LAN to be able to decode
> signals such as faxes. Can I configure a linux workstation to do due
> this through a sound card and a HF receiver ? ALso can I set up a work

I have some HF fax code for linux, though it would require some
polishing. See http://ife.ee.ethz.ch/~sailer/ham/linux/fax.html.

Then there was a posting on linux-hams by DL4SDC, which I'm appending.

Tom

35. REMOVE

From: Farrag Adel - Electronics Eng <afa@staffmail.rtc-tallaght.ie>
Date: Thu, 13 Aug 1998 09:30:37 +0100
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <5720F6784D8BD1119F2A00A0C94BC69116F480@GPO.it-tallaght.ie>
Rendered body type: text/plain

-----Original Message-----
        From: Nathan Sykes [SMTP:nsykes@bellsouth.net]
        Sent: 11 August 1998 05:57
        To: hfsig@tapr.org
        Subject: [HFSIG:2732] remove

        -----Original Message-----
        From: knowit98@hotmail.com <knowit98@hotmail.com>

To: hfsig@tapr.org <hfsig@tapr.org>
Date: Monday, August 10, 1998 9:19 PM
Subject: [HFSIG:2731] View Amazing World Record Sex!!!

>Attention!
>Warning! Adults Only! Warning! Adults Only!
>
>If you are under 21 years of age, or not interested in sexually explicit material...
>please hit your keyboard delete button now
>and please excuse the intrusion. To REMOVE your name from our
>email list send us email with REMOVE in the subject line.
>You need not read any further!
>
>
>Available NOW for only $9.95! Next 10 Days Only!
>
>WORLD RECORD SEX!
>
>Be There! See It Now On Video! Unbelievable ...But True!
>
>You Won't Believe Your Eyes!!! [As Seen on the Howard Stern Show]
>
>"The World's Biggest Gang Bang"
>
>See sexy Annabel Chong as she sets the world Gang Bang Record
>in this fantastic video documentary that chronicles her 24 hour
>sexathon with 251 men engaging in sexual intercourse and oral
>sex with her! Don't worry, you won't have to stay up 24 hours
>to watch it all. We've selected only the most exciting and red
>hot scenes for you...all in breathtaking living color with plenty of
>extreme close-ups! This video is guaranteed to knock your socks
>off and leave you breathless! You've never seen anything like it!
>Annabel takes on five men at a time! 90 minutes! Order Today!
>Only $9.95 plus $3 shipping and handling [Total $12.95].
>
>
>"GANG BANG II"
>The Record Breaker!!! Starring Jasmin St. Claire!
>
>See Beautiful and Voluptious Jasmin St. Claire shatter Annabel's
>gang bang record by taking on 300 men in one 24 hour sex session!
>You won't believe your eyes at all the hot firey action that you will
>see as the new world record is established before your eyes as
>Jasmin takes on five men at a time for sexual intercourse and
>oral sex! Your friends will break down your door to see this video!
>You'll be the most popular guy in town! The action is truly unreal
>and you will see the best of it in living life-like color! Order Today
>and see Jasmin break the record! 90 minutes. Only $9.95 plus
>$3 shipping and handling [total $12.95].
>
>Also Available...
>
>The Uncensored Authentic Underground...
>Pamela Anderson Lee & Tommy Lee
>Sex Video Tape!
>
>Everyone is talking about this exciting video! See Pam and
>Tommy engaging in sexual intercourse and oral sex in the car,
>on the boat and much, much more! A real collectors video! 30 minutes.
>Only $9.95 plus $3 shipping and Handling [total $12.95]
>
>"Tonya Harding Wedding Night Sex Video"
>Now see the beautiful Ice Skating Shame of the Olympics
>Tonya Harding engaging in sexual intercourse and oral

>sex on her wedding night with husband Jeff Gillooly!
>This "Bad Girl" is Hot! Don't miss this video! 30 minutes.
>Only $9.95 plus $3 shipping and handling [total $12.95]
>
>"Traci...I Love You" Starring Traci Lords
>Now see the most beautiful and popular porn star in her last
>adult video before she hit the big time! It's the blockbuster
of
>the year...sensual...fiery and exposive! Traci Lords in her
most
>erotic and controversial film ever! Don't Miss It! 90
minutes.
>Only $9.95 plus $3 shipping and handling [total $12.95]
>
>EMAIL SPECIAL!
>
>ORDER ANY FOUR VIDEOS AND GET THE FIFTH ONE FREE!!!
>
>Your order will be shipped via First Class Mail. All Shipments
in plain
unmarked wrapper.
>For Priority Mail - Add $5
>For Overnight Express - add $15
>
>You can order by Phone, Fax, Mail or Email.
>
>We accept all Major Credit Cards and checks by phone or fax.
>Visa - MasterCard - American Express - Discover
>
>10 Day Money Back Guarantee! We know that you will be pleased
with these
Videos!
>
>To Email your order - DO NOT HIT REPLY ON YOUR KEYBOARD
>
>Send email to our special email address below:
>
>infoplus467@juno.com
>
>[Note: If you order by email and do not receive an email
acknowledgement
within 24 hours, please phone our office at
>718-287-3800]
>
>Phone our office 9am to 10 pm [eastern time]
>
>[718] 287-3800 to Order By Phone for FASTEST SERVICE!
>
> We can accept your credit card or check by phone
>
>Fax Your Order 24 hours per day to [718] 462-5920
>You can fax your credit card information or your check
>
>Order by mail by sending $12.95 per video, cash, check, money
order or
major credit card [Visa, MasterCard, American Express or
Discover] to
>
>TCPS, INC.
>4718 18th Ave. Suite 135
>Brooklyn, NY 11204
>
>Make Checks & Money Orders Payable to TCPS, Inc.
>New York State Residents Please Add 85 cents for Sales Tax per
Video!
>
>You must be over 21 years of age to order and give us your date
of birth
with your order!
>
>The Following Order Form is for Your Convenience!
>

>.....
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>
>
>

>
>Please ship me the following video tape[s]!
>
>Qty___________Annabel Chong "World's Biggest Gang Bang"
>
>Qty__________"Gang Bang II" Jasmin St. Claire
>
>Qty___________"Pamela & Tommy Lee Sex Video Tape"
>
>Qty_________   "Tonya Harding Wedding Night Sex Video Tape"
>
>Qty__________"Traci I Love You" Traci Lords
>
>at $9.95 each plus $3.00 for shipping and handling per tape
>[$12.95 per video or "SPECIAL $51.80 for ALL FIVE"!
>
>Credit Card #______________________________Exp Date___
>
>I hereby represent that I am over 21 years of age.
>My date of birth is_________________________________
>
>Signature____
>
>Ship to: Name_______________________________________
>
>Address____
>
>City________________________State___________Zip________
>
>Area Code and Home Phone [   ]___________________________
>
>Fax # [   ]____
>
>Email Address____
>
>To remove your name from our mailing list, send us an email
with
>remove in the subject line. This is a one time offer and you
should not
hear from us again!
>
>FOREIGN ORDERS -Add $15us if you desire Air Parcel Post
Shipment. We ship
all over the world.
>
>By deleting your unwanted E-Mail you waste one keystroke, yet
>by throwing away paper mail you waste our planet! SAVE THE
>TREES and support internet E-Mail instead of paper mail!
>
>[C] Copyright TCPS 1998
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36. Re: [HFSIG:2751] Any HF data activity?

From: Mike Bernock <mbernock@mediaone.net>
Date: Thu, 13 Aug 1998 08:26:33 -0400
To: hfsig@tapr.org
Message-ID: <35D2DB79.CD0ACC4A@mediaone.net>
Rendered body type: text/plain

Gary Chatters wrote:
> Is there anyone on this list actually doing HF data experiments?

Gary,

Yes. I am one of three hams in the northeast who are active in
evaluating HF digital modes via over-the-air testing (KB1JY, W1IMM,
KB1PZ - me). So far we have taken a look at packet, AMTOR, PACTOR, GTOR,
and CLOVER II, and have published several papers. We are all located
within 100 mi. of Boston so our testing has emphasized near vertical
incidence skywave (NVIS) paths.

73s,

-- Mike, KB1PZ
mbernock@mediaone.net

37. Re: [APRSSIG:27140] APRS on CW!

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Thu, 13 Aug 1998 09:48:11 -0400 (EDT)
To: aprssig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <Pine.SOL.3.96.980813090757.4311B-100000@arctic>
Rendered body type: text/plain

> On Wed, 12 Aug 1998, Bill Vodall wrote:
>
> > Anybody ever made a CW-APRS gateway? It wouldn't be hard.

Great Idea for a project!. It would be easy to do, and all you do
is convert the CW to a packet on 144.39 where it would be picked up
automatically by the nearest APRS Internet Gatewway for worldwide
distribution! Thus ANYONE, ANYWHERE with a CW key or the preverbial two
wires in an emergency, could send (and receive) his POSITION and MESSAGES
to/from anywhere in the world!

of the FOUR basic APRS FORMATS, only the POSIT/COMMENT and the MESSAGE
format are needed. THese could be handled in CW as:

```
      TOCALL de WB4APR Message........            MESSAGE
      QTH    de WB4APR LLLLN/LLLLLWcomment...     POSIT LAT/LONG
      GRID   de WB4APR GGggGG..........           Grid Square posit
```

The APRS/CW gateway would ack all correctly understood "packets"
with a (SN) ...-. followed by a single letter T, Q, or G to show that it
parsed the packet into one of the three formats...

WOW! This is the perfect keep-CW-alive project for someone. APRS has
given a great shot in the arm for packet. It could do this for CW to!

AND just like all the internet gateways are bidirectional, the
APRS/CW gateway could be also! Whenever it sees on the worldwide APRS
internet stream, a message for ANYONE who has recently been heard on CW,
it would turn around and send it OUT on CW!

THUS worldwide APRS two-way comms from anywhere to anywhere as long as you
have the proverbial CW or two wires to click together. NO TNC! Just a
radio!

de WB4APR, BOB...

38. RE: [HFSIG:2762] Re: [APRSSIG:27140] APRS on CW!

From: "Lanier, Robert" <rlanier@harris.com>
Date: Thu, 13 Aug 1998 10:27:16 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <275399FB18C4D111871300805FBEB72F01215AF7@CORPMX6>
Rendered body type: text/plain

Right on, Bob!!

> -----Original Message-----
> From: hfsig@tapr.org [SMTP:hfsig@tapr.org]
> Sent: Thursday, August 13, 1998 8:52 AM
> To: hfsig@tapr.org
> Subject: [HFSIG:2762] Re: [APRSSIG:27140] APRS on CW!
>
> > On Wed, 12 Aug 1998, Bill Vodall wrote:
> >
> > > Anybody ever made a CW-APRS gateway? It wouldn't be hard.
>
> Great Idea for a project!. It would be easy to do, and all you do
> is convert the CW to a packet on 144.39 where it would be picked up
> automatically by the nearest APRS Internet Gatewway for worldwide
> distribution! Thus ANYONE, ANYWHERE with a CW key or the preverbial
two
> wires in an emergency, could send (and receive) his POSITION and
MESSAGES
> to/from anywhere in the world!
>
> of the FOUR basic APRS FORMATS, only the POSIT/COMMENT and the MESSAGE
> format are needed. THese could be handled in CW as:
>
> TOCALL de WB4APR Message........ MESSAGE

```
              QTH    de WB4APR LLLLN/LLLLLWcomment...     POSIT LAT/LONG
              GRID   de WB4APR GGggGG..........           Grid Square posit

        The APRS/CW gateway would ack all correctly understood "packets"
        with a (SN) ...-.  followed by a single letter T, Q, or G to show that
it
        parsed the packet into one of the three formats...

        WOW!  This is the perfect keep-CW-alive project for someone.  APRS has
        given a great shot in the arm for packet.  It could do this for CW to!

        ***AND*** just like all the internet gateways are bidirectional, the
        APRS/CW gateway could be also!  Whenever it sees on the worldwide APRS
        internet stream, a message for ANYONE who has recently been heard on CW,
        it would turn around and send it OUT on CW!

        THUS worldwide APRS two-way comms from anywhere to anywhere as long as
you
        have the proverbial CW or two wires to click together.  NO TNC! Just a
        radio!

        de WB4APR, BOB...
```

39. Re: [HFSIG:2757] Human readable digital modes

```
From: dona@tridelta.com (Don Adams)
Date: Thu, 13 Aug 1998 10:46:12 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199808131446.KAA29709@sparcy.tridelta.com>
Rendered body type: text/plain
```

This looks interesting.....however it took me a while to find this site.
The real address is http://members.xoom.com/ZL1BPU/Contents.html

Murry could you send me the GWBasic program to generate the tones?

Also I'm just playing around with my receiver, using the Icom serial
interface in DOS (or DOS window) Quick Basic. Does anyone
have an FFT routine in somesort of street basic or C?. I wrote one
years ago that would convert any N points by factoring in ALGOL
(Does anyone remenber that language?) but if anyone has a QB
program, it would be quicker for me to get started.

>To enrich your experience of Amateur Radio, check out the site:
>
>http://members.xoom/ZL1BPU/Contents.html
>
>73,
>
>Murray ZL1BPU
>
>

40. Re: [HFSIG:2761] Re: Any HF data activity?

```
From: dona@tridelta.com (Don Adams)
Date: Thu, 13 Aug 1998 10:57:30 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199808131457.KAA29804@sparcy.tridelta.com>
Rendered body type: text/plain
```

>
 So far we have taken a look at packet, AMTOR, PACTOR, GTOR,
>and CLOVER II, and have published several papers.
>-- Mike, KB1PZ
>mbernock@mediaone.net

Are these papers on the net? Or where can we get a copy........

over......Don wa8qzz

41. Re: [HFSIG:2761] Re: Any HF data activity?

```
From: Anthony Martin <anthony.martin@gecm.com>
Date: Thu, 13 Aug 1998 16:23:35 +0100
To: hfsig@tapr.org
Message-ID: <35D304F7.47B0@gecm.com>
Rendered body type: text/plain
```

Mike Bernock wrote:
>

> and CLOVER II, and have published several papers. We are all located
> within 100 mi. of Boston so our testing has emphasized near vertical
> incidence skywave (NVIS) paths.
>
NVIS is an interesting area, many HF radio systems are being used in
an NVIS mode. It offers higher SNR than longer skip distances but
still has the multipath. It has the potential to offer much higher
data rates with modem designs appopriately matched to NVIS channels.
14400 bits/s & higher would not be unreasonable on NVIS.

I've suggested that my company develop higher datarate HF modems
specifically for NVIS but without much success. I think there
would be a market but I'm not the marketing man.

Looks like I may be working on 8Mbits/s troposcatter in November.
Currently our longest 8 megabit tropo link is 500km at around 900MHz.

Anthony

42. Re: [HFSIG:2762] Re: [APRSSIG:27140] APRS on CW!

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 13 Aug 1998 12:33:05 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.980813123003.9106B-100000@www.kelly-afb.org>
Rendered body type: text/plain

I at one time modified a Packet BBS program that ran on BASIX and a Radio
Shack M100 to run on CW. I used an AEA PK-232 in the CW mode. I just
couldn't get much local interest on 10M to test it out before putting it
up on 40 or 75 meters. Put almost any of the old Basic BBS programs can
be re-written to accept CW.

Walt

```
=============================================================================
|           In a world without fences, who needs Gates?                     |
|---------------------------------------------------------------------------|
|        If I wanted "Windows", I'd knock a hole in my wall.                |
=============================================================================
|                                  |                                        |
|                                  | The Micro$oft operating system         |
| Walt DuBose - K5YFW              | didn't get as bad as it is             |
| E-Mail k5yfw@www.kelly-afb.org   | overnight.  I has taken over 10        |
| Business Telephone: (210)925-1171| years of careful, calculated,          |
|     Home Telephone: (210)696-3196| development.                           |
|                                  |                                        |
=============================================================================
|   Using MS is like eating high class dog food...its still dog food.       |
=============================================================================
```

43. Re: [HFSIG:2751] Any HF data activity?

From: w2pat@juno.com (Marvin Bernstein)
Date: Wed, 12 Aug 1998 20:24:13 -0400
To: hfsig@tapr.org
Message-ID: <19980813.155956.7430.0.w2pat@juno.com>
Rendered body type: text/plain

Hi Gary;
     I have been running some experiments on HF, using both G-tor and
Pactor. To avoid some problems with QRM, I have operated on an AF Mars
frequency with other Members. The results of some of these tests were
reported in the PSK, published by TAPR. Recently I completed the final
report on a series of tests in which changes were made in the Filters and
Demodulators, using the Pactor Mode. I have started a new series of tests
using Amtor, in which I am trying to determine the benfits of Pass Band
Tuning and the W9GR DSP RTTY Filter. If you are interested, I will keep
you informed on any progress in these experiments. I have operated rtty
for many years but now use only the new data modes. 73.
Marv
w2pat/afa1da

_____
You don't need to buy Internet access to use free Internet e-mail.
Get completely free e-mail from Juno at http://www.juno.com
Or call Juno at (800) 654-JUNO [654-5866]

44. remove

From: Jim Sanford <jsanfor@exis.net>
Date: Thu, 13 Aug 1998 16:55:04 -0400
To: hfsig@tapr.org
Message-ID: <35D352A8.8962A19B@exis.net>
Rendered body type: text/plain

knowit98@hotmail.com wrote:

> Attention!
> Warning! Adults Only!   Warning! Adults
>   Only!
>
> If you are under 21 years of age, or not interested in sexually explicit material...
> please hit your keyboard delete button now
> and please excuse the intrusion. To REMOVE your name from our
> email list send us email with REMOVE in the subject line.
> You need not read any further!
>
> Available NOW for only $9.95! Next 10 Days Only!
>
> WORLD RECORD SEX!
>
> Be There! See It Now On Video! Unbelievable ...But True!
>
> You Won't Believe Your Eyes!!! [As Seen on the Howard Stern Show]
>
> "The World's Biggest Gang Bang"
>
> See sexy Annabel Chong as she sets the world Gang Bang Record
> in this fantastic video documentary that chronicles her 24 hour
> sexathon with 251 men engaging in sexual intercourse and oral
> sex with her! Don't worry, you won't have to stay up 24 hours
> to watch it all. We've selected only the most exciting and red
> hot scenes for you...all in breathtaking living color with plenty of
> extreme close-ups! This video is guaranteed to knock your socks
> off and leave you breathless! You've never seen anything like it!
> Annabel takes on five men at a time! 90 minutes! Order Today!
> Only $9.95 plus $3 shipping and handling [Total $12.95].
>
> "GANG BANG II"
> The Record Breaker!!! Starring Jasmin St. Claire!
>
> See Beautiful and Voluptious Jasmin St. Claire shatter Annabel's
> gang bang record by taking on 300 men in one 24 hour sex session!
> You won't believe your eyes at all the hot firey action that you will
> see as the new world record is established before your eyes as
> Jasmin takes on five men at a time for sexual intercourse and
> oral sex! Your friends will break down your door to see this video!
> You'll be the most popular guy in town! The action is truly unreal
> and you will see the best of it in living life-like color! Order Today
> and see Jasmin break the record! 90 minutes. Only $9.95 plus
> $3 shipping and handling [total $12.95].
>
> Also Available...
>
> The Uncensored Authentic Underground...
> Pamela Anderson Lee & Tommy Lee
> Sex Video Tape!
>
> Everyone is talking about this exciting video! See Pam and
> Tommy engaging in sexual intercourse and oral sex in the car,
> on the boat and much, much more! A real collectors video! 30 minutes.
> Only $9.95 plus $3 shipping and Handling [total $12.95]
>
> "Tonya Harding Wedding Night Sex Video"
> Now see the beautiful Ice Skating Shame of the Olympics
> Tonya Harding engaging in sexual intercourse and oral
> sex on her wedding night with husband Jeff Gillooly!
> This "Bad Girl" is Hot! Don't miss this video! 30 minutes.
> Only $9.95 plus $3 shipping and handling [total $12.95]
>
> "Traci...I Love You" Starring Traci Lords
> Now see the most beautiful and popular porn star in her last
> adult video before she hit the big time! It's the blockbuster of
> the year...sensual...fiery and exposive! Traci Lords in her most
> erotic and controversial film ever! Don't Miss It! 90 minutes.
> Only $9.95 plus $3 shipping and handling [total $12.95]
>
> EMAIL SPECIAL!
>

> ORDER ANY FOUR VIDEOS AND GET THE FIFTH ONE FREE!!!
>
> Your order will be shipped via First Class Mail. All Shipments in plain unmarked wrapper.
> For Priority Mail - Add $5
> For Overnight Express - add $15
>
> You can order by Phone, Fax, Mail or Email.
>
> We accept all Major Credit Cards and checks by phone or fax.
> Visa - MasterCard - American Express - Discover
>
> 10 Day Money Back Guarantee! We know that you will be pleased with these Videos!
>
> To Email your order - DO NOT HIT REPLY ON YOUR KEYBOARD
>
> Send email to our special email address below:
>
> infoplus467@juno.com
>
> [Note: If you order by email and do not receive an email acknowledgement within 24 hours, please phone
 our office at
> 718-287-3800]
>
> Phone our office 9am to 10 pm [eastern time]
>
> [718] 287-3800 to Order By Phone for FASTEST SERVICE!
>
> We can accept your credit card or check by phone
>
> Fax Your Order 24 hours per day to [718] 462-5920
> You can fax your credit card information or your check
>
> Order by mail by sending $12.95 per video, cash, check, money order or major credit card [Visa,
MasterCard, American Express or Discover] to
>
> TCPS, INC.
> 4718 18th Ave. Suite 135
> Brooklyn, NY 11204
>
> Make Checks & Money Orders Payable to TCPS, Inc.
> New York State Residents Please Add 85 cents for Sales Tax per Video!
>
> You must be over 21 years of age to order and give us your date of birth with your order!
>
> The Following Order Form is for Your Convenience!
>
> ....
.....
>
> Please ship me the following video tape[s]!
>
> Qty_____Annabel Chong "World's Biggest Gang Bang"
>
> Qty_____"Gang Bang II" Jasmin St. Claire
>
> Qty_____"Pamela & Tommy Lee Sex Video Tape"
>
> Qty_____   "Tonya Harding Wedding Night Sex Video Tape"
>
> Qty_____"Traci I Love You" Traci Lords
>
> at $9.95 each plus $3.00 for shipping and handling per tape
> [$12.95 per video or "SPECIAL $51.80 for ALL FIVE"!
>
> Credit Card #_____Exp Date___
>
> I hereby represent that I am over 21 years of age.
> My date of birth is_____
>
> Signature_____
>
> Ship to: Name_____
>
> Address_____
>
> City_____State_____Zip_____
>
> Area Code and Home Phone [   ]_____
>

> Fax # [   ]____
>
> Email Address____
>
> To remove your name from our mailing list, send us an email with
> remove in the subject line. This is a one time offer and you should not hear from us again!
>
> FOREIGN ORDERS -Add $15us if you desire Air Parcel Post Shipment. We ship all over the world.
>
> By deleting your unwanted E-Mail you waste one keystroke, yet
> by throwing away paper mail you waste our planet! SAVE THE
> TREES and support internet E-Mail instead of paper mail!
>
> [C] Copyright TCPS 1998

45. Re: [HFSIG:2761] Re: Any HF data activity?

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 13 Aug 1998 18:31:19 -0800
To: hfsig@tapr.org
Message-ID: <199808140134.SAA20422@peak.org>
Rendered body type: text/plain

>Gary Chatters wrote:
> Is there anyone on this list actually doing HF data experiments?

I might add that in the past few weeks we have had very nice round
table QSO's between myself in Oregon, Jim, K6OYY in Santa Barara,
Les, VK2DSG in Australia, and Fred OH/DK4ZC in Finland on 20 meters.

We are using both the MTHELL and MT63 modes. MT63 especially
works extremely well even under the weakest of band conditions, though
I must admit that conditions are not bad at the moment.

Incidentally, if you don't know what these modes are: MTHELL is a
frequency-domain FAX mode, and MT63 is a multitone modem .. packs
63 tones in 1200 Hz with FEC and deep interleaving. Les, VK2DSG
often plays back our transmissions and it's really impressive seeing the
results of using only 40W output power and a vertical antenna.
This all is running on a Motorola EVM.

Trust this will encourage further interest.

--Johan, KC7WW

46. Re: [HFSIG:2759] Re: Radio and Network set up

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 13 Aug 1998 18:38:24 -0800
To: hfsig@tapr.org
Message-ID: <199808140141.SAA21505@peak.org>
Rendered body type: text/plain

Tom,

>Then there was a posting on linux-hams by DL4SDC, which I'm appending.

Very interesing -- thanks for pointing this out. Do you know
whether Andreas has uploaded the code somewhere for downloading yet?

73

Johan, KC7WW

47. Re: [HFSIG:2771] Re: Radio and Network set up

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Fri, 14 Aug 1998 16:18:50 +0200
To: hfsig@tapr.org
Message-ID: <35D4474A.B7642E80@ife.ee.ethz.ch>
Rendered body type: text/plain

Johan Forrer wrote:

> Very interesing -- thanks for pointing this out. Do you know
> whether Andreas has uploaded the code somewhere for downloading yet?

No, unfortunately not, but I haven't asked him. There were some
suggestions
on the linux-hams list where he could put it but I don't know what

happened
after that.

Tom

48. REMOVE

From: "harens" <harens@netcom.ca>
Date: Mon, 17 Aug 1998 21:00:39 -0400
To: <hfsig@tapr.org>
Message-ID: <003b01bdca43$9e86dce0$64aefea9@home>
Rendered body type: text/plain

[Empty message body]

49. Re: [HFSIG:2765] Re: Any HF data activity?

From: Mike Bernock <mbernock@mediaone.net>
Date: Mon, 17 Aug 1998 21:58:20 -0400
To: hfsig@tapr.org
Cc: kwick@mitre.org
Message-ID: <35D8DFBC.B2D9BDB6@mediaone.net>
Rendered body type: text/plain

To Don and the rest of the group,

I have forwarded your replies to KB1JY (Ken, ad-hoc test Direktor and
author of most published works) who seems to have lost his subscription
to this mailing list. Publications which contain some of our work are:

PACKET: Speed, more Speed and Applications, 2nd Edition ARRL Publication
(test results for AMTOR, PACTOR, GTOR, packet over NVIS and OHS links,
and the results of some HF ALE testing)
and
QEX, December '96 (KA9Q NOS TCP/IP over NVIS links)

I expect Ken to pipe up here shortly with some follow up info and
possibly some invites to those interested in participating in future
testing.

-- Mike, KB1PZ

50. HF-NVIS and ALE

From: Patricia Gibbons <wa6ube@ix.netcom.com>
Date: Tue, 18 Aug 1998 19:13:27 -0700
To: hfsig@tapr.org
Message-ID: <35DA34C6.7EB2@ix.netcom.com>
Rendered body type: text/plain

I am interested in networking
with others who may be
exploring the use of HF-NVIS
propagation modes for HF data
communications via the use of
equipment that is also capable
of ALE.

In the next week I will be in
the process of setting up a new
/\/\otorola MICOM-2R military/ruggedized
set for Pactor/G-Tor operation on
frequencies between 2.5 and 10 MHz.

Please reply to: <wa6ube@ix.netcom.com>

Thanx

Trish

--

Patricia E. Gibbons
Acting Chief Communications Technician
City of San Jose - ITD/communications
....
My Public Key is available at:
<http://pgp5.ai.mit.edu/pks-commands.html>
Key ID: 0xEDECB44F
This key is RSA, NOT Diffie-Hellman !!

51. Re: [HFSIG:2775] HF-NVIS and ALE

From: "B.Haineault" <bhaineau@cts.com>
Date: Tue, 18 Aug 1998 20:32:57 -0700
To: <hfsig@tapr.org>
Message-ID: <008101bdcb22$0fd0a320$d805a3cd@bhaineau.cts.com>
Rendered body type: text/plain

Patricia,

I could coordinate a similar set-up at my end... but San Diego --> San Jose
is not a "NVIS" path !
If it were, we could have used some of our experimental HF frequencies to
run ALE/NVIS data tests.

Keep me posted.

Bruno Haineault, KF6QDP/VE2EQ
Director, Engineering Services Group
DATRON
San Marcos, California

ICQ Pager #: 335336
ICQ Pager address: http://wwp.mirabilis.com/335336

-----Original Message-----
From: Patricia Gibbons <wa6ube@ix.netcom.com>
To: hfsig@tapr.org <hfsig@tapr.org>
Date: Tuesday, 1998 August, 18 19:39
Subject: [HFSIG:2775] HF-NVIS and ALE

>I am interested in networking
>with others who may be
>exploring the use of HF-NVIS
>propagation modes for HF data
>communications via the use of
>equipment that is also capable
>of ALE.
>
>In the next week I will be in
>the process of setting up a new
>/\/\otorola MICOM-2R military/ruggedized
>set for Pactor/G-Tor operation on
>frequencies between 2.5 and 10 MHz.
>
>Please reply to: <wa6ube@ix.netcom.com>
>
>Thanx
>
>Trish
>
>--
>
>Patricia E. Gibbons
>Acting Chief Communications Technician
>City of San Jose - ITD/communications
>.....................................
>My Public Key is available at:
><http://pgp5.ai.mit.edu/pks-commands.html>
>Key ID: 0xEDECB44F
>This key is RSA, NOT Diffie-Hellman !!
>
>

52. Re: [HFSIG:2775] HF-NVIS and ALE

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Wed, 19 Aug 1998 00:31:26 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.96.980819002954.13890H-100000@arctic>
Rendered body type: text/plain

On Tue, 18 Aug 1998, Patricia Gibbons wrote:

> I am interested in networking with others who may be
> exploring the use of HF-NVIS propagation modes for HF data
> communications via the use of equipment that is also capable
> of ALE.

I was one of the 100 who bought the surplus 39 tone HARRIS 2400 baud HF

modems. TO date I ahve not had any time to play. Has anyone else?

de WB4APR

53. Re: [HFSIG:2775] HF-NVIS and ALE

From: Charles Brain <chbrain@dircon.co.uk>
Date: Wed, 19 Aug 1998 08:53:30 +0100
To: hfsig@tapr.org
Message-ID: <35DA847A.AB62AD4B@dircon.co.uk>
Rendered body type: text/plain

Patricia Gibbons wrote:
>
> equipment that is also capable
> of ALE.
>
> In the next week I will be in
> the process of setting up a new
> /\/\otorola MICOM-2R military/ruggedized
> set for Pactor/G-Tor operation on
> frequencies between 2.5 and 10 MHz.
>
> Please reply to: <wa6ube@ix.netcom.com>
>
> Thanx
>
> Trish
>
Well I suppose I must final admit what I have been up to!

I have written a Windows 95 implementation of the ALE protocols.
It is in a very early Beta state at the moment and requires
at least a 100 Mhz Pentium and a SoundCard. The 8ary modem
being implemented using the Sound Card.

As I said it is very experimental. But does receive
ALE traffic and will establish links, I have tried it with a couple
of other controllers so I know the protocols are correct.
The parts I am a bit suspicious about are the slotted responses
in NET and GROUP calls and the ARQ DTM and DBM protocols. AMD is
fine.

<<This is for experimentors only who have time to tinker.>>

Although I have it installed and working on two of my machines
including a Toshiba Laptop, someone else has had difficulty in
getting it to work due to missing DLLs on their machine etc
(you have been warned).

I have also disabled the linking protection (encryption) to make it legal.

Contact me if anyone is interested.

- Charles G4GUO

54. Re: [HFSIG:2775] HF-NVIS and ALE

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Wed, 19 Aug 1998 07:28:00 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.980819071144.3667A-100000@www.kelly-afb.org>
Rendered body type: text/plain

Trish, et al,

I worked with ALE for a number of years while in the AF Reserves and
Chief of Communications for the Aeromedical Evacuation System. We used
primarily Harris military equipment but also used Rockwell-Collins and
Motorola equipment.

While is Saudi, during Desert Shield/Storm, we tried to use ALE on our
Aeromedical Evacuation Net. This was 8-12 frequencies between 3.3 and 15
MHz. Typically we didn't use the lowest and highest frequencies in our
channelization...thus generally used only 10 channels.

What we discovered was that if there were communications being carried
out on a certain channel (QSO going on) the ALE "stepped on" the
communications in progress in its attempt to make a connection with the
desired station.

On the amateur bands, this would be considered willful interference...as the ALEs we had then did _NOT_ listen for a "QSO" on frequency or near the frequency before transmitting their ALE signals.

If ALEs don't listen on a frequency (channel) to insure that it is not in use, then using ALE on the Amateur Frequencies could be considered a violation of the FCC rules should the ALE signal on top of, or interfere with a QSO in progress.

I might also mention here that we used the Harris Serial tone modem to transmit 4800 BSP on HF at S:N ratios less that 2:1. The Harris parallel modem that Bob and others bought should be able to send data at a rate of 2400 BSP at nearly the same SNR and perhaps 4800 BPS at SNR of 5:1. The Harris parallel tone model is described in a MIL STD that I have in my files and others on this mail reflection also have. I am told that this modem _CAN_ be duplicated using the Motorola DVM. Most hams running packet of one sort or the other on HF have a hard time of getting data thruputs of more that 2400 BPS for the above stated SNR...much less 4800 BPS and this is still better than most hams have on V/UHF...its still normally 1200 BPS up there.

Walt/K5YFW


```
    ========================================================================
    |           In a world without fences, who needs Gates?                |
    |----------------------------------------------------------------------|
    |        If I wanted "Windows", I'd knock a hole in my wall.           |
    ========================================================================
    |                                  |                                   |
    |                                  | The Micro$oft operating system    |
    | Walt DuBose - K5YFW              | didn't get as bad as it is        |
    | E-Mail k5yfw@www.kelly-afb.org   | overnight.  I has taken over 10   |
    | Business Telephone: (210)925-1171 | years of careful, calculated,    |
    |     Home Telephone: (210)696-3196 | development.                     |
    |                                  |                                   |
    ========================================================================
    |   Using MS is like eating high class dog food...its still dog food.  |
    ========================================================================
```


On Tue, 18 Aug 1998, Patricia Gibbons wrote:

> I am interested in networking
> with others who may be
> exploring the use of HF-NVIS
> propagation modes for HF data
> communications via the use of
> equipment that is also capable
> of ALE.
>
> In the next week I will be in
> the process of setting up a new
> /\/\otorola MICOM-2R military/ruggedized
> set for Pactor/G-Tor operation on
> frequencies between 2.5 and 10 MHz.
>
> Please reply to: <wa6ube@ix.netcom.com>
>
> Thanx
>
> Trish
>
> --
>
> Patricia E. Gibbons
> Acting Chief Communications Technician
> City of San Jose - ITD/communications
> ....
> My Public Key is available at:
> <http://pgp5.ai.mit.edu/pks-commands.html>
> Key ID: 0xEDECB44F
> This key is RSA, NOT Diffie-Hellman !!
>
>

55. Re: [HFSIG:2779] Re: HF-NVIS and ALE

From: Charles Brain <chbrain@dircon.co.uk>
Date: Wed, 19 Aug 1998 17:00:18 +0100
To: hfsig@tapr.org

Message-ID: <35DAF692.DA24EA87@dircon.co.uk>
Rendered body type: text/plain

Walt DuBose - K5YFW wrote:

> On the amateur bands, this would be considered willful interference...as
> the ALEs we had then did _NOT_ listen for a "QSO" on frequency or near the
> frequency before transmitting their ALE signals.
>

This is the principal reason why I have given up on MIL-STD ALE and
have curtailed my development efforts. I tried to add extra DSP software
to detect Voice transmissions on the channel before transmission, but it
was so unreliable as to be next to useless. Probably my implementation!

I never considered it suitable for Amateur Radio Operation anyhow. About the only
thing I thought it might be suitable for was an HF APRS system where messages
are very short.

A military system I worked on used the same modem for traffic as for call
establishment so as to reduce mutual interference with circuits in progress.

I have also seen a couple of other systems that use central frequency planning tools
to try to avoid this problem and they probably do to a certain extent. But this
would not be possible in the Amateur Service.

It should be possible to produce an ALE noise blanker as the wave form is so
simple. There is a project for someone!

Anyway Some of the BBS system currently fielded by Amateurs are effectively simple
ALE systems by the fact they scan a number of channels looking for calls. If you disable
the LQA database of a 188-141A controller that is in effect what you get.

When you start to make Group or Net calls choosing the right channel becomes
very difficult. One controller I saw even required you to manually enter the frequency
you were going to call on!

So don't worry Walt I don't think anyone would be stupid enough to use
ALE on the HAM bands in anger. Maybe outside :)

> modem _CAN_ be duplicated using the Motorola DVM. Most hams running

Yes it can be done there is easily enough MIPS.

I have now shifted my interest into the use of DSP in direction finding.

- Charles G4GUO

56. UNSUBSCRIBE HFSIG

From: Arvydas Vaisnys <arvydas.vaisnys@jpl.nasa.gov>
Date: Wed, 19 Aug 1998 10:33:47 -0700
To: hfsig@tapr.org
Message-ID: <3.0.32.19980819103346.008c0c10@zorba.jpl.nasa.gov>
Rendered body type: text/plain

>From: listproc@tapr.org
>Date: Fri, 14 Aug 1998 11:13:10 -0500 (CDT)
>Reply-To: listproc@tapr.org
>Sender: listproc@tapr.org
>To: Arvydas.Vaisnys@jpl.nasa.gov
>Cc: tapr-mail@tapr.org
>Subject: UNSUBSCRIBE HFSIG
>X-Listprocessor-Version: 6.0 -- ListProcessor by Anastasios Kotsikonas
>X-Comment: Tucson Amateur Packet Radio list processor
>
>You have been removed from list hfsig@tapr.org.
>Thanks for being with us.
>
I am still with you, only now I am getting one copy of each message instead
of two.
This is certainly an improvement.

Arv

57. HF radio used by Municipal govt ??

From: "Patricia E. Gibbons" <wa6ube@ix.netcom.com>
Date: Thu, 20 Aug 1998 10:34:48 -0700
To: lmr@ucsd.edu, hfsig@tapr.org
Message-ID: <35DC5E38.742B@ix.netcom.com>

Rendered body type: text/plain

Our Fire Department is exploring options
for city-wide radio coverage within the
City of San Jose if our dispatch center
and all of our existing communicaitons
infrastructure were off the air due to
a major disaster..

Has anyone heard of the ability of local
Govt at municipal level to successfully
get licensed on a few, perhaps three
channels in the HF spectrum in order
to use HF radio and NVIS operations ??

For what our Fire Department would like
it terms of coverage it would look like
the three freq's would be near

2.5 MHz, 4.5 MHz, and another in the
6 Mhz range.

I know that the Central Gov'ts of each
State has this ability and each county
participates with the State Govt in
HF radio net operations, but I have not
run across such a system for municipal
use by a large City such as City of Los
Angeles or in our case, City of San Jose.

Anyone else run across this ??

Thanx

Trish
--

Patricia E. Gibbons
City of San Jose - ITD/communications
Acting Chief Communications Technician
My Public Key is available at:
<http://pgp5.ai.mit.edu/pks-commands.html>
Key ID: 0xEDECB44F
This key is RSA, NOT Diffie-Hellman !!

58. Re: [HFSIG:2782] HF radio used by Municipal govt ??

From: dona@tridelta.com (Don Adams)
Date: Thu, 20 Aug 1998 15:48:12 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <199808201948.PAA23845@sparcy.tridelta.com>
Rendered body type: text/plain

Trish ,
I've worked with this sort of problem recently.
To make a long story short, in general people
go back to VHF LO band 30-50 Mhz. With todays
equipment, I would say you could have a 100mile
diameter area. ie 50 mile or more radius. You can
run 100watts or more from a mobile. And several
hundred from from a base.
The antennas at these frequencie, 45 mhz, are relatively
efficent vs 40 80 or 160 meters. I wouldn't
like to go below 7mhz. The antenna there
has maybe a few kilohertz of bandwidth.
At 2.5 mhz, it's really narrow. ~10 to 50khz

over......Don wa8qzz

>Our Fire Department is exploring options
>for city-wide radio coverage within the
>City of San Jose if our dispatch center
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>infrastructure were off the air due to
>a major disaster..
>
>Has anyone heard of the ability of local
>Govt at municipal level to successfully
>get licensed on a few, perhaps three
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>

>--
>
>Patricia E. Gibbons
>City of San Jose - ITD/communications
>Acting Chief Communications Technician
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><http://pgp5.ai.mit.edu/pks-commands.html>
>Key ID: 0xEDECB44F
>This key is RSA, NOT Diffie-Hellman !!
>
>

59. Re: [HFSIG:2783] Re: HF radio used by Municipal govt ??

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 20 Aug 1998 15:10:48 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.980820150726.8502J-100000@www.kelly-afb.org>
Rendered body type: text/plain

I have to agree. I used to work the 30-50 MHz commercial band in the
South Texas oil fields back in the late 50's and early 60's. A 100 watt
FM mobile with 1/4 wave whips can talk between 20-30 miles most of the
time. Base stations with simple antennas at 30-50 ft can talk 30-50 miles.

How big is San Jose?

Walt

```
    ========================================================================
    |           In a world without fences, who needs Gates?                |
    |----------------------------------------------------------------------|
    |        If I wanted "Windows", I'd knock a hole in my wall.           |
    ========================================================================
    |                                  |                                   |
    |                                  | The Micro$oft operating system    |
    | Walt DuBose - K5YFW              | didn't get as bad as it is        |
    | E-Mail k5yfw@www.kelly-afb.org   | overnight.  I has taken over 10   |
    | Business Telephone: (210)925-1171 | years of careful, calculated,    |
    |     Home Telephone: (210)696-3196 | development.                     |
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    ========================================================================
    |   Using MS is like eating high class dog food...its still dog food.  |
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```

On Thu, 20 Aug 1998, Don Adams wrote:

> Trish ,
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> To make a long story short, in general people
> go back to VHF LO band 30-50 Mhz. With todays
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> diameter area. ie 50 mile or more radius. You can
> run 100watts or more from a mobile. And several
> hundred from from a base.
> The antennas at these frequencie, 45 mhz, are relatively
> efficent vs 40 80 or 160 meters. I wouldn't
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> has maybe a few kilohertz of bandwidth.
> At 2.5 mhz, it's really narrow. ~10 to 50khz
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> over......Don wa8qzz
>
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> >for city-wide radio coverage within the
> >City of San Jose if our dispatch center
> >and all of our existing communicaitons
> >infrastructure were off the air due to
> >a major disaster..
> >

> >Has anyone heard of the ability of local
> >Govt at municipal level to successfully
> >get licensed on a few, perhaps three
> >channels in the HF spectrum in order
> >to use HF radio and NVIS operations ??
> >
> >For what our Fire Department would like
> >it terms of coverage it would look like
> >the three freq's would be near
> >
> >2.5 MHz, 4.5 MHz, and another in the
> >6 Mhz range.
>
> >
>
> >--
> >
> >Patricia E. Gibbons
> >City of San Jose - ITD/communications
> >Acting Chief Communications Technician
> >My Public Key is available at:
> ><http://pgp5.ai.mit.edu/pks-commands.html>
> >Key ID: 0xEDECB44F
> >This key is RSA, NOT Diffie-Hellman !!
> >
> >
>
>
>

60. Re: [HFSIG:2782] HF radio used by Municipal govt ??

From: "Louis Cobet" <louis_cobet@email.msn.com>
Date: Thu, 20 Aug 1998 20:42:20 -0700
To: <hfsig@tapr.org>
Message-ID: <007301bdccb5$b42c7040$7b242599@MSNLouisCobet>
Rendered body type: text/plain

NVIS at the lower frequencies you mention should cover a large part of the state of California. Almost all of San Jose can be covered by a 20 mile radius circle. Public service frequencies in the 40 Mhz. range should work fine.

Louis Cobet, K6MDH
louis_cobet@email.msn.com

You wrote
 >Our Fire Department is exploring options
>for city-wide radio coverage within the
>City of San Jose if our dispatch center
>and all of our existing communicaitons
>infrastructure were off the air due to
>a major disaster..
>
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>
>I know that the Central Gov'ts of each
>State has this ability and each county
>participates with the State Govt in
>HF radio net operations, but I have not
>run across such a system for municipal
>use by a large City such as City of Los
>Angeles or in our case, City of San Jose.
>
>Anyone else run across this ??
>
>Thanx
>
>Trish

>--
>
>Patricia E. Gibbons
>City of San Jose - ITD/communications
>Acting Chief Communications Technician
>My Public Key is available at:
><http://pgp5.ai.mit.edu/pks-commands.html>
>Key ID: 0xEDECB44F
>This key is RSA, NOT Diffie-Hellman !!
>

61. Re: Fwd: Re: Fwd: Re: [HFSIG:2782] HF radio used by Municipal govt ??

From: Patricia Gibbons <wa6ube@ix.netcom.com>
Date: Fri, 21 Aug 1998 20:16:36 -0700
To: "Paul E. Robbins" <Paul.E.Robbins@jpl.nasa.gov>, hfsig@tapr.org,
lmr@ucsd.edu
Cc: Don_Root/OES@oes.ca.gov
Message-ID: <35DE3814.480C@ix.netcom.com>
Rendered body type: text/plain

Paul E. Robbins wrote:
>
> HI Trish, I took the liberty of forwarding your message up to John Peterson
> at Ames Research Center, note please his comments. John is in emergency
> preparedness at Ames as well as the Ames Spectrum Manager. --some---
> ---snipping---
> Regards, Paul Robbins JPL Spectrum Management Specialist
>

Hello Paul and thanx for the info...

I should have been more explicit on my original posting.
because I wasn't, I am posting this amplified response
back to the hfsig mailing list.

What the Fire department wants to do is only equip our
five battalion chiefs with radios such that they can
communicate, radio to radio, without the need of any
other infrastructure and use this system "When the
big one happens". The intent was not to outfit
every Fire Vehicle with HF equipment.

They are looking for a command frequency such that they
can all communicate, intercom, party-line-fashion,
and not be dependant on any existing infrastructure to
allow them to communicate. My best guess is that there
<might> be some nook or crannie of the "SHARES" program
where they would fit, but I think that for them to even
fit in that context, it would have to be part of a more regional
system like the entire SF-Bay Area. It seems really
spectrum inefficient for one municipality to get licensed
on even three HF channels in the 2.0-6.0 MHz range.
Then again, I think thats why SHARES was created.

The terrain of Municipal San Jose has a footprint
large enough that our infrastructure for our normal
operations require 20+ radio sites, voting receivers,
and repeater systems with multiple transmitter sites.
Most of our sites are all low-level except for one.
even our one high-level site (Mt. Chual) will not cover
many parts of Almaden valley, South San Jose, and other
new development areas. Our City, as I recall, is now
over one million people and has surpassed the City
and County of San Francisco in population size and
square miles.

Pardon my ignorance, but I am simply exploring any
available option that might do the job when the chips
are down. So far, the best alternative seems to be a
low-band VHF frequency like down in the 33 MHz range.

We have an EOC-to-EOC net down there now that is
County-wide on 37.08 MHz, and another command-net on low-band
that ties each of the Fire Dispatch centers together,
and yet another for each of the Police Department
Dispatch facilities, so perhaps the end result is
just licensing 6 or 7 mobiles on the existing
"Red NET", Low-band system if everyone is really
willing to put up with intra-agency traffic on

what is supposed to be a county-wide inter-agency
system. It boils down to a matter of politics and
philosophy as to whether a net would include not
just dispatch centers but Senior command staff
existing in the field and trying to efficiently
share the channel.

My gosh, we're going to end up going full
circle --> we are now in the middle of
"narrow-banding" our radios for "spectrum
refarming" just like in the 1960's with
"split-channel", and expanding
systems on low-band at the same time
 EEEEK!!   =:-0

Thanx

Trish
--

Patricia E. Gibbons
Acting Chief Communications Technician
City of San Jose - ITD/communications
....
My Public Key is available at:
<http://pgp5.ai.mit.edu/pks-commands.html>
Key ID: 0xEDECB44F
This key is RSA, NOT Diffie-Hellman !!

62. New platforms available for PSK31

From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
Date: Mon, 24 Aug 1998 14:21:33 +0200
To: hfsig@tapr.org, dsp@tapr.org
Message-ID: <3.0.5.32.19980824142133.00842b20@bicc00.bi.ehu.es>
Rendered body type: text/plain

The summertime has been fruitfull and I am pleased to announce new
versions/modems of PSK31:

First,

        There is a full Windows version for the Motorola EVM by Peter Martinez
that can transmit 255 different varicode symbols, and maps these to values
in the range 0 to 255 for display as characters on the screen. This means
that PSK31 is capable of handling some of the less-often-used symbols and
special accented characters that are not in the standard 128-character
ASCII character set. Note, however, that the character-mapping used is the
ANSI standard, intended for world-wide use and not the character mapping
used by DOS programs, which is different for each country and depends on
the code page installed in your computer's display and keyboard drivers.

        This version supports automatic loading of code on the EVM if the PLXBOOT
code is loaded on the EEPROM and is a deligth to operate.

        It's available here: ftp://det.bi.ehu.es/pub/ham/psk31/psk31evmw.zip

Second,

        There is a version of PSK31 for the Analog Devices by Michael Keller
DL6IAK. You will need to perform an small modification (soldering a wire)
on the SHARC board but it's compatible will all other software for it.

        This version available only for Windows, has the unique capability to have
a spectral display to help finding the frequency.

        It's available here:

        ftp://det.bi.ehu.es/pub/ham/psk31/psk31shcw.zip

        As a reminder, the official PSK31 Webpage is in

        http://aintel.bi.ehu.es/psk31.html

        Best regards and hoping to meet you soon on PSK31.

        Eduardo/EA2BAJ

Eduardo Jacob - Area de Ingenieria Telematica

```
Departamento de Electronica y Telecomunicaciones
ETSII y de IT               Tel: +34-(9)4-427 8055
UPV / EHU                   Fax: +34-(9)4-441 4041
Alda Urquijo s/n         E-mail: jtpjatae@bi.ehu.es
E-48013 - Bilbao (Spain)
PGP Key available: Send email w/subject: PGPKEY
fingerprint      : D8 FA 54 49 F7 40 BA AA 09 1A 73 40 A4 26 ED BD
```

63. MTHell and FeldHell for the EVM

```
From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
Date: Mon, 24 Aug 1998 14:30:44 +0200
To: hfsig@tapr.org, dsp@tapr.org
Message-ID: <3.0.5.32.19980824143044.00858380@bicc00.bi.ehu.es>
Rendered body type: text/plain
```

After a Beta test time, I am pleased to tell you that there are two new packages by Peter G3PLX that implement a Windows Feldhell terminal and a Windows MTHell (Multitone Hell) terminal. Both are for the Motorola EVM and available here:

http://det.bi.ehu.es/pub/ham/evm/Hell/feldhell.zip
http://det.bi.ehu.es/pub/ham/evm/Hell/mthell.zip

Regards and hoping to do DX on MT/FeldHELL

Eduardo/EA2BAJ

```
---------------------------------------------------------
Eduardo EA2BAJ
DSP and HAM:   http://det.bi.ehu.es/~jtpjatae/ham.html
VHF PACKET: EA2BAJ @ EA2URV
PGP Key avlb: Send email w/subject: PGPKEY
fingerprint : D8 FA 54 49 F7 40 BA AA 09 1A 73 40 A4 26 ED BD
```

64. Re: [psk31] New platforms available for PSK31

```
From: "Douglas & Nadia Braun" <dougbert@otenet.gr>
Date: Mon, 24 Aug 1998 21:19:46 +0300
To: <hfsig@tapr.org>, <dsp@tapr.org>,
        "Eduardo Jacob" <jtpjatae@bicc00.bi.ehu.es>
Message-ID: <199808241819.VAA01838@athserv.otenet.gr>
Rendered body type: text/plain
```

Thanks for the information. By the way, I will be moving from Greece back to the USA in a couple of weeks, and all my gear is being packed up right now.
Hopefully in a couple of months I wil be back on the air...

Doug
SV/N1OWU

```
----------
> From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
> To: hfsig@tapr.org; dsp@tapr.org
> Subject: [psk31] New platforms available for PSK31
> Date: 24 August 1998 15:21
>
> The summertime has been fruitfull and I am pleased to announce new
> versions/modems of PSK31:
```

65. Re: [[HFSIG:2788] MTHell and FeldHell for the EVM]

```
From: Alberto di Bene <dibene@usa.net>
Date: 25 Aug 98 00:15:04 MET DST
To: hfsig@tapr.org
Message-ID: <19980824221504.29117.qmail@www0g.netaddress.usa.net>
Rendered body type: text/plain
```

Eduardo, I think that the correct URLs should be:

  ftp://det.bi.ehu.es/pub/ham/evm/Hell/feldhell.zip
  ftp://det.bi.ehu.es/pub/ham/evm/Hell/mthell.zip

and not:

  http://det.bi.ehu.es/pub/ham/evm/Hell/feldhell.zip
  http://det.bi.ehu.es/pub/ham/evm/Hell/mthell.zip

73,
Alberto, I2PHD

_____
Get free e-mail and a permanent address at http://www.netaddress.com/?N=1

66. Re: [HFSIG:2790] Re: [MTHell and FeldHell for the EVM]

From: Eduardo Jacob <edu@kender.es>
Date: Tue, 25 Aug 1998 07:46:07 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19980825074607.00868a00@kender.es>
Rendered body type: text/plain

At 17:18 24/08/98 -0500, you wrote:
>Eduardo, I think that the correct URLs should be:
>
> ftp://det.bi.ehu.es/pub/ham/evm/Hell/feldhell.zip
> ftp://det.bi.ehu.es/pub/ham/evm/Hell/mthell.zip
>
>and not:
>
> http://det.bi.ehu.es/pub/ham/evm/Hell/feldhell.zip
> http://det.bi.ehu.es/pub/ham/evm/Hell/mthell.zip

Yes I just noticed it. When you publish using Netscape, it seems it automagically sets the URL to http, and I copied it from the page. On the Page, they where at least well referenced.

Eduardo

September 1998

Original source file: hfsig.9809
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9809

1. Human Readable Modes

From: "M. Greenman" <coombedn@ihug.co.nz>
Date: Tue, 1 Sep 1998 22:42:38 +1200
To: <hfsig@tapr.org>
Message-ID: <199809011042.WAA31857@smtp2.ihug.co.nz>
Rendered body type: text/plain

Hi,
The "Fuzzy Digital Modes" web site,

http://members.xoom.com/ZL1BPU/Contents.html

is dedicated to Feld-Hell and Multi-Tone Hell, and now carries the
following new high performance software:

Feld - Hell for EVM by Peter G3PLX
16 tone Concurrent MT-Hell for EVM by G3PLX
Feld-Hell for SB16 sound card by Lionel G3PPT
7 tone Concurrent MT-Hell for SB16 sound card by G3PPT
7 tone Sequential MT-Hell for PC speaker output by ZL1BPU - new version
with BIG print option
(Use GRAM spectrograph program to receive these modes - see the web site)

European DX on MT-Hell is there for the taking! Try 14063.5 kHz LSB at
0430Z Saturday and Sunday.

SEE you in Hell!

Murray ZL1BPU

---

2. Re: Error Condition Re: unsubscribe

From: DTalaska@aol.com
Date: Wed, 2 Sep 1998 23:49:21 EDT
To: hfsig@tapr.org
Message-ID: <3411e922.35ee11c1@aol.com>
Rendered body type: text/plain

dtalaska
unsubscribe dtalaska@aol.com

---

3. Re: Error Condition Re: unsubscribe

From: DTalaska@aol.com
Date: Wed, 2 Sep 1998 23:49:20 EDT
To: hfsig@tapr.org
Message-ID: <411438ed.35ee11c0@aol.com>
Rendered body type: text/plain

dtalaska
unsubscribe dtalaska@aol.com

---

4. Re: [HFSIG:2793]

From: DTalaska@aol.com
Date: Thu, 3 Sep 1998 11:38:11 EDT
To: hfsig@tapr.org
Message-ID: <3f1a86ce.35eeb7e3@aol.com>
Rendered body type: text/plain

dtalaska
unsubscribe dtalaska@aol.com

---

5. Re: [HFSIG:2794] Re: Error Condition Re: unsubscribe

From: Sexylex11@aol.com
Date: Thu, 3 Sep 1998 21:48:07 EDT
To: hfsig@tapr.org
Message-ID: <38e31239.35ef46d7@aol.com>
Rendered body type: text/plain

please stop sending me this mail!!! I will report you if you don't. I remind
you that this is a warning!

---

6. RE: [HFSIG:2796] Re: Error Condition Re: unsubscribe

From: "Lanier, Robert" <rlanier@harris.com>
Date: Fri, 4 Sep 1998 09:11:33 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <275399FB18C4D111871300805FBEB72F01215B2B@corpmx6.ess.harris.com>
Rendered body type: text/plain

What's this all about?

Tony KE4ATO

-----Original Message-----
From: hfsig@tapr.org [SMTP:hfsig@tapr.org]
Sent: Thursday, September 03, 1998 8:50 PM
To: hfsig@tapr.org
Subject: [HFSIG:2796] Re: Error Condition Re: unsubscribe

please stop sending me this mail!!! I will report you if you don't. I remind
you that this is a warning!

7. Re: [HFSIG:2792] Human Readable Modes

From: Eduardo Jacob <edu@kender.es>
Date: Sat, 05 Sep 1998 16:04:31 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19980905160431.007ebab0@kender.es>
Rendered body type: text/plain

>European DX on MT-Hell is there for the taking! Try 14063.5 kHz LSB at
>0430Z Saturday and Sunday.

It's very early but I'll try to be there. Let me know if you don't plan to be there...

Eduardo/EA2BAJ

8. Re: [HFSIG:2798] Re: Human Readable Modes

From: Daisym3090@aol.com
Date: Sat, 5 Sep 1998 10:35:36 EDT
To: hfsig@tapr.org
Message-ID: <c92e985c.35f14c38@aol.com>
Rendered body type: text/plain

What?

9. Re: [HFSIG:2799] Re: Human Readable Modes

From: Daisym3090@aol.com
Date: Sat, 5 Sep 1998 11:05:02 EDT
To: hfsig@tapr.org
Message-ID: <8a6e1fb4.35f1531e@aol.com>
Rendered body type: text/plain

You told me to meet you somewhere or something, I didn't know what you meant.
Have I talked to you before?

10. Re: [HFSIG:2798] Re: Human Readable Modes

From: Kirsten369@aol.com
Date: Sat, 5 Sep 1998 11:16:00 EDT
To: hfsig@tapr.org
Message-ID: <9433d324.35f155b0@aol.com>
Rendered body type: text/plain

Who are you?

11. Re: [HFSIG:2800] Re: Human Readable Modes

From: Kirsten369@aol.com
Date: Sat, 5 Sep 1998 11:38:57 EDT
To: hfsig@tapr.org
Message-ID: <5fff2a34.35f15b11@aol.com>
Rendered body type: text/plain

I don't even know who u r? r u a girl or guy? When did I tell u this?

12. Re: [HFSIG:2801] Re: Human Readable Modes

From: Kirsten369@aol.com
Date: Sat, 5 Sep 1998 11:40:02 EDT
To: hfsig@tapr.org
Message-ID: <c33d3234.35f15b52@aol.com>
Rendered body type: text/plain

In a message dated 9/5/98 10:37:40 AM Central Daylight Time,
Kirsten369@aol.com writes:

<<
 Who are you?
  >>
I want to know who u r first and when i said that...cause I don't think I did

13. Re: Hi

From: Turtle1283@aol.com
Date: Sat, 5 Sep 1998 11:49:59 EDT
To: hfsig@tapr.org
Message-ID: <178609f5.35f15da7@aol.com>
Rendered body type: text/plain

Please tell me how to unsubscribe from your list. I don't want to be sent any
of this e-mail any more. thanx.

14. Re: [HFSIG:2804] Re: Hi

From: Kirsten369@aol.com
Date: Sat, 5 Sep 1998 12:30:02 EDT
To: hfsig@tapr.org
Message-ID: <77f37042.35f1670a@aol.com>
Rendered body type: text/plain

what list

15. Re: [HFSIG:2801] Re: Human Readable Modes

From: Daisym3090@aol.com
Date: Sat, 5 Sep 1998 12:53:38 EDT
To: hfsig@tapr.org
Message-ID: <50906d02.35f16c92@aol.com>
Rendered body type: text/plain

Who ARE YOU? Why do you keep sending me this shit I don't WANT IT OK? you
bitch

16. Re: [HFSIG:2805] Re: Hi

From: Daisym3090@aol.com
Date: Sat, 5 Sep 1998 13:02:49 EDT
To: hfsig@tapr.org
Message-ID: <cd0a7d86.35f16eb9@aol.com>
Rendered body type: text/plain

STOP TALKING TO ME DAMN IT! YOU BITCH

17. Re: [HFSIG:2806] Re: Human Readable Modes

From: Kirsten369@aol.com
Date: Sat, 5 Sep 1998 14:02:48 EDT
To: hfsig@tapr.org
Message-ID: <be73da4e.35f17cc8@aol.com>
Rendered body type: text/plain

I'm not sending you any mail

18. Re: [HFSIG:2807] Re: Hi

From: Kirsten369@aol.com
Date: Sat, 5 Sep 1998 14:03:10 EDT
To: hfsig@tapr.org
Message-ID: <8e45c7e9.35f17cde@aol.com>
Rendered body type: text/plain

I'm not talking to you

19. Re: [HFSIG:2798] Re: Human Readable Modes

From: mail <jon@agari.org>
Date: Sat, 05 Sep 1998 15:03:01 -0400
To: hfsig@tapr.org
Message-ID: <35F18AE5.9921F969@agari.org>
Rendered body type: text/plain

Is it possible to translate this using a PC soundcard and a good shortwave if you have
the proper software ?

Eduardo Jacob wrote:

> >European DX on MT-Hell is there for the taking! Try 14063.5 kHz LSB at
> >0430Z Saturday and Sunday.

>
>   It's very early but I'll try to be there. Let me know if you don't plan to
> be there...
>
>   Eduardo/EA2BAJ
>

20. Re: [HFSIG:2807] Re: Hi

From: Kim51184@aol.com
Date: Sat, 5 Sep 1998 18:01:29 EDT
To: hfsig@tapr.org
Message-ID: <71fd3bb0.35f1b4b9@aol.com>
Rendered body type: text/plain

um excuse me but i don't know you so stop sending junk got it!!!!

21. (no subject)

From: "Calvin Keli Lunny" <lunny@dave-world.net>
Date: Sat, 5 Sep 1998 17:02:44 -0500
To: <hfsig@tapr.org>
Message-ID: <01bdd918$e8e7f4c0$f46d19ce@lunny.dave-world.net>
Rendered body type: text/plain

signoff hfsig

22. Re: [HFSIG:2811] Re: Hi

From: Daisym3090@aol.com
Date: Sat, 5 Sep 1998 18:20:42 EDT
To: hfsig@tapr.org
Message-ID: <2fe4e63c.35f1b93a@aol.com>
Rendered body type: text/plain

I haven't been sending you mail

23. [HFSIG:2813] Re: Hi

From: PRNCESSMYA@aol.com
Date: Sat, 5 Sep 1998 19:04:26 EDT
To: hfsig@tapr.org
Message-ID: <242892cc.35f1c37a@aol.com>
Rendered body type: text/plain

listen stop sending me this mail please i don`t know you or them peoples
that`s in the e-mails

24. Re: [HFSIG:2811] Re: Hi

From: Lucymuffet@aol.com
Date: Sat, 5 Sep 1998 20:23:42 EDT
To: hfsig@tapr.org
Message-ID: <c85854a5.35f1d60e@aol.com>
Rendered body type: text/plain

hi Idont knoww you either kiss my
asssssssssssssssssssssssshole!!!!!!!!!!!!!!!!!!!!!!!

25. Re: [HFSIG:2815] Re: Hi

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Sat, 5 Sep 1998 20:11:48 -0600
To: hfsig@tapr.org
Message-ID: <v04011702b2179fba6beb@[207.43.172.70]>
Rendered body type: text/plain

Have no fear. All the various SPAM aol.com addresses that got added have
been removed.

Please return to the normal HF SIG discussions...and ignore these messages.

Greg

Greg Jones, WD5IVD   Austin, Texas
wd5ivd@tapr.org   http://www.tapr.org/~wd5ivd

26. Re: [HFSIG:2807] Re: Hi

From: KIMBA1LOV@aol.com
Date: Sat, 5 Sep 1998 21:45:05 EDT
To: hfsig@tapr.org
Message-ID: <9f9b8955.35f1e921@aol.com>

Rendered body type: text/plain

hey you jackass i dont know what your problem is but you dont need to be
sending me anymore nasty email ok thank you very much

27. Re: [HFSIG:2809] Re: Hi

From: KIMBA1LOV@aol.com
Date: Sat, 5 Sep 1998 21:47:14 EDT
To: hfsig@tapr.org
Message-ID: <40fded5.35f1e9a2@aol.com>
Rendered body type: text/plain

hey what have i ever done to you to make you not want to talk to me
please tell me i havnt the faintest clue

28. Re: [HFSIG:2811] Re: Hi

From: KIMBA1LOV@aol.com
Date: Sat, 5 Sep 1998 21:50:11 EDT
To: hfsig@tapr.org
Message-ID: <2d412e56.35f1ea53@aol.com>
Rendered body type: text/plain

dont know what your problem is but i never sent you any junk mail thank you
very much i dont even know who you are

29. TNX, OM (Was: Re: [HFSIG:2816] Re: Hi)

From: Gary Chatters <gc@universe.digex.net>
Date: Sat, 5 Sep 1998 22:33:00 -0400
To: hfsig@tapr.org
Message-ID: <199809060233.WAA08875@universe.digex.net>
Rendered body type: text/plain

>Have no fear. All the various SPAM aol.com addresses that got added have
>been removed.

Thank you.

Z^3

30. Human Readable Modes

From: "M. Greenman" <coombedn@ihug.co.nz>
Date: Sun, 6 Sep 1998 16:51:06 +1200
To: <hfsig@tapr.org>
Cc: <jon@agari.org>
Message-ID: <199809060450.QAA09030@smtp2.ihug.co.nz>
Rendered body type: text/plain

This is in reply to an HFSIG message from jon@agari.org that read:

Is it possible to translate this using a PC soundcard and a good shortwave
if you have
the proper software ?

It was in reply to a note from Eduardo EA2BAJ about the skeds on 14063.5

The answer is YES, but you do need special software. With GRAM, obtainable
at

http://www.monumental.com/rshorne/gram.html

and a 16 bit sound card, you can read all the various multi-tone
Hellschreiber modes. For an explanation of these modes, see the site:

http://members.xoom.com/ZL1BPU/Contents.html

There are other signals on this same frequency which you will not be able
to read - I am listening now at 0448Z and they are transmitting PSK31
instead. Be patient....

Murray ZL1BPU

31. Re: [HFSIG:2821] Human Readable Modes

From: Eduardo Jacob <edu@kender.es>
Date: Sun, 06 Sep 1998 19:25:38 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19980906192538.00867100@kender.es>
Rendered body type: text/plain

At 00:09 6/09/98 -0500, you wrote:
>It was in reply to a note from Eduardo EA2BAJ about the skeds on 14063.5

Well, I was there but I didn't get anything on that freq. Incidentally I was able at that time to get ZL6B Beacon clearly... :-(I was willing to get a QSO with my antipodes (I live in one of those few sites with antipodes on land)

Eduardo

32. Re Human readable modes!

From: "Stan Symons" <stan@g3dss.demon.co.uk>
Date: Sat, 5 Sep 1998 03:12:42 +1
To: hfsig@tapr.org
Message-ID: <E0zFlAC-0000EM-00@post.mail.demon.net>
Rendered body type: text/plain

I was listening on 14063.5 this morning using the MTHELL program and was able to copy DK4ZC/OH and VK2DKP without too much difficulty. I also managed a few words from ZL1BDV.

Look fwd to more tests. Most encouraged by my first efforts.

73 de stan G3DSS

33. 80/40 meter Gtor/Pactor & AOL'rs attached to hfsig list

From: Patricia Gibbons <wa6ube@ix.netcom.com>
Date: Sun, 06 Sep 1998 23:18:29 -0700
To: hfsig@tapr.org
Message-ID: <35F37AB5.3940@ix.netcom.com>
Rendered body type: text/plain

WOW! was that an earfull of stuff from
people who don't understand mailing list
servers!! :-(

As for other topics:

Is anyone here on the west coast that is
active on 80 and 40 meter Pactor/Gtor ??

Seems like most people are never on 40
or 80 meters..

I've been setting up on 7082 KHz
(7081 mark tone freq) during the daytime
and on 3624 Khz (3623 mark-tone) in the evenings.

Anyone want to do a sked ??

Trish
--

Patricia E. Gibbons
Acting Chief Communications Technician
City of San Jose - ITD/communications
.....
My Public Key is available at:
<http://pgp5.ai.mit.edu/pks-commands.html>
Key ID: 0xEDECB44F
This key is RSA, NOT Diffie-Hellman !!

34. Re: [HFSIG:2811] Re: Hi

From: Jamie24g@aol.com
Date: Mon, 7 Sep 1998 15:31:01 EDT
To: hfsig@tapr.org
Message-ID: <4cad26aa.35f43475@aol.com>
Rendered body type: text/plain

um excuse me but i don't know you so why would i send you joke mail? get a clue

35. Re: [HFSIG:2813] Re: Hi

From: Jamie24g@aol.com
Date: Mon, 7 Sep 1998 15:33:19 EDT
To: hfsig@tapr.org
Message-ID: <888b52b.35f434ff@aol.com>
Rendered body type: text/plain

hi
i have gotten many letters like yours someone is sending mail in my name don't know how but they are. because you see i'm not even here i am in florida, this is my mom typing this message will try to fix the problem

sorry for any trouble

36. Amateur radio with ALE

From: Patricia Gibbons <wa6ube@ix.netcom.com>
Date: Mon, 07 Sep 1998 14:46:15 -0700
Cc: hfsig@tapr.org
Message-ID: <35F45427.570B@ix.netcom.com>
Rendered body type: text/plain

Is there anyone out there that would like to experiment with
ALE on the ham bands ?? I've noticed that my new rig, which
has Automatic Link Establishment, has an option in the programming
to not sound on an occupied channel. This seems like it may work
in the ham bands if the later versions of ALE are polite in the
respect of not sending out ALE broadcasts on a channel that is
busy.

I would like to hear from other hams that may have equipment,
surplus or otherwise, that has ALE implemented in it's design,
especially if you are located on the West Coast of the US.

Trish
--

Patricia E. Gibbons
Acting Chief Communications Technician
City of San Jose - ITD/communications
.....
My Public Key is available at:
<http://pgp5.ai.mit.edu/pks-commands.html>
Key ID: 0xEDECB44F
This key is RSA, NOT Diffie-Hellman !!

37. Human Readable Modes, MT-Hell

From: murrayg@indcom.co.nz
Date: Tue, 8 Sep 1998 10:24:08 +1200
To: hfsig@tapr.org
Message-ID: <199809072216.RAA22092@tapr.org>
Rendered body type: text/plain

Hi,
As you folk will have gathered, I worked Fred OH/DK4ZC for the first
time on Sunday, just a short while after I posted my previous note!

A first ZL - EU MT-Hell contact!

14063.5 kHz is the place to be. Saturday and Sunday from 0430 onward
you will hear PSK31 and MT63, but with patience, (and a bit of
encouragement via SSB!) the other folk will send MT-Hell for you. You
should hear OH/DK4ZC, VK2DSG and possibly one or more DL/DF
stations.

Stan G3DSS and Eduardo, thanks for listening - keep trying, because as
winter approaches your conditions should get better. I recognise that the
limitation is reception conditions in Europe. A better time might be 0600 or
0630, not so early for you, and the others will be almost finished by then.

Copy at this end is generally very good. I can email samples of EU signals
to anyone who wants. If you want me to print your transmissions, let me
know. You should be able to work VK2DSG easily as he has a good
location and a big antenna. I'll be listening!

By the way, I'm claiming a world's first sequential MT-Hell contact for 1st
July 1998 with ZL1AN. Anybody think they have a prior claim?

Murray ZL1BPU
as149@detroit.freenet.org

38. unsubscribe

From: Lynn Coffelt <lynnc@sos.net>
Date: Mon, 07 Sep 1998 18:05:55 -0700
To: hfsig@tapr.org
Message-ID: <35F482F1.3181AE79@sos.net>

Rendered body type: text/plain

Greg:
Please, please help me unsubscribe from hfsig. I've tried other
avenues, but with no luck.
                        Thanks in advance, Lynn, W7LTQ since 1948.
Avid radio enthusiast, but this is too agrivating for me!

39. hi

From: Turtle1283@aol.com
Date: Mon, 7 Sep 1998 22:59:38 EDT
To: hfsig@tapr.org
Message-ID: <1e5bfe81.35f49d9a@aol.com>
Rendered body type: text/plain

Please tell me how to unsubscribe from your list. I don't want to be sent any
of this e-mail from HFSIG:2804 any more. thanx.

40. Remove

From: Oris <oris@dirtball.com>
Date: Mon, 07 Sep 1998 23:47:08 -0400
To: hfsig@tapr.org
Message-ID: <35F4A8BC.4A90@dirtball.com>
Rendered body type: text/plain

[Empty message body]

41. Psk31 for Linux and SoundBlaster

From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
Date: Fri, 11 Sep 1998 16:54:07 +0200
To: psk31@bipt106.bi.ehu.es
Message-ID: <3.0.5.32.19980911165407.0083c240@bicc00.bi.ehu.es>
Rendered body type: text/plain

Hansi Reiser DL9RDZ has prepared a PSK31 version for Linux and the
SoundBlaster.

        It's available at

        ftp://det.bi.ehu.es/pub/ham/psk31/psk31_lx.tgz

        btw, linux style, only sources, no binary...

        As usual, you can take a look at what PSK31 is at it's homepage:

        http://aintel.bi.ehu.es/psk31.html

        Regards

        Eduardo

PD: Now if somebody want to write a DOS or Windows version with SB, he has
part of the work done.

Eduardo EA2BAJ
DSP and HAM: http://det.bi.ehu.es/~jtpjatae/ham.html
VHF PACKET: EA2BAJ @ EA2URV
PGP Key avlb: Send email w/subject: PGPKEY
fingerprint : D8 FA 54 49 F7 40 BA AA 09 1A 73 40 A4 26 ED BD

42. TAPR.ORG Down and now Back

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Mon, 14 Sep 1998 21:09:36 -0500
To: "HF SIG list   mailing"<hfsig@tapr.org>, " Spread Spectrum "<ss@tapr.org>,
        " TAPR/AMSAT DSP "<dsp@tapr.org>,
        "APRS SIG list   mailing"<aprssig@tapr.org>,
        "NETSIG list   mailing"<netsig@tapr.org>,
        "BBS SIG list   mailing"<bbssig@tapr.org>,
        " tacgps "<tacgps@tapr.org>, aprsnews, mic-e,
        TAPR Regional Freq <regional_freq@tapr.org>
Message-ID: <v0401170fb2237bbfafc8@[207.8.125.53]>
Rendered body type: text/plain

TAPR.ORG was down on Saturday for a system upgrade. We installed a new
processor board as well as added more memory and a new 9G HD.

Sunday morning very early, the root drive crashed and burned. Lee spent
Sunday and today (Monday) recovering from the crash on root. We had backed
up the system just before the upgrade, so we didn't get hit very hard. We

took this time to also upgrade the OS on the system, which we had planned
to in the future.

Lee got the system back operational about 6pm tonight and all aspects of
the system should be operational by this posting. We will be spending the
rest of tonight doing checks on various automatic operations to make sure
that everything is running correctly.

Cheers - Greg

Greg Jones, WD5IVD
        President TAPR (http://www.tapr.org)
        PhD Candidate, Instructional Technology, UT Austin

email: wd5ivd@tapr.org
web: http://www.tapr.org/~wd5ivd

Experience is what you get when you were expecting something else.

43. 1998 ARRL and TAPR DCC this coming weekend !

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Tue, 22 Sep 1998 17:41:08 -0500
To: "TAPR-BB list   mailing"<tapr-bb@tapr.org>
Cc: "HF SIG list   mailing"<hfsig@tapr.org>, " Spread Spectrum "<ss@tapr.org>,
        " TAPR/AMSAT DSP "<dsp@tapr.org>,
        "APRS SIG list   mailing"<aprssig@tapr.org>,
        "NETSIG list   mailing"<netsig@tapr.org>,
        "BBS SIG list   mailing"<bbssig@tapr.org>,
        " tacgps "<tacgps@tapr.org>, aprsnews, mic-e,
        TAPR Regional Freq <regional_freq@tapr.org>
Message-ID: <v04011717b22dd78686a9@[128.83.74.103]>
Rendered body type: text/plain

Just as a reminder the 1998 ARRL and TAPR Digital Communications Conference
in Chicago, IL is happening this weekend!

The conference schedule and information is on-line at http://www.tapr.org/dcc
The abstract page should be on-line shortly.

Hope to see you in Chicago this weekend!

Cheers - Greg Jones, WD5IVD

Greg Jones, WD5IVD   Austin, Texas
wd5ivd@tapr.org   http://www.tapr.org/~wd5ivd

44. newbie test and questions

From: W6RCecilA <w6rca@ibm.net>
Date: Fri, 25 Sep 1998 13:40:27 -0700
To: hfsig@tapr.org
Message-ID: <360BFFBB.3158@ibm.net>
Rendered body type: text/plain

This is my first entry to see if it works. If it does, I
am recently retired from Intel and have some questions.

1. What is the status of Pentium (I&II) MMX digital applications
software using nothing but a PC + sound board?

2. Have we or others standardized on a particular PC sound board?

3. Is there applications support driver SW for such a sound board?

4. Is the DSP56002EVM Motorola Eval Board a good choice for
TAPR supported experimentation? In other words, will I get $240
worth of good out of it if I buy one and join the experimentation
group?

There's nothing wrong with state-of-the-art complex systems.
What about a very simple digital system that could compete with
and replace CW? How about an AMTOR-like ASCII based variable-
baud-rate FSK system?

Simple system suggestion: Use the built-in FSK transmitting
capabilities of my ICOM-745 and ICOM-706 and decode the FSK
with a PC and a sound board - no TNC or Eval board required.
Make it auto-baud-rate variable all the way down to CW's 12wpm
equivalent of 10 baud.

My email address is W6RCA@ibm.net. I've recently retired from
Intel after designing DSP technology into the MCS-x96 line of
µcontrollers. I know how to do IIR and FIR filters with
µcontrollers. Anyone feel free to drop me an email. I've got
a lot of catching up to do regarding TAPR stuff.
--
73, Cecil, W6RCA   http://people.delphi.com/CecilMoore

45. INCREASE SALES UP TO 200% - REFERRAL PROGRAM - $50-$150 PER REFERRAL

From: cdcyl@usa.net
Date: Mon, 28 Sep 1998 04:15:29 +0100 (BST)
To: cdcyl@usa.net
Message-ID: <m0zNTmb-0004S4C@corpex.com>
Rendered body type: text/plain

WIN

        WISCONSIN OR OREGON WILDLIFE PARADISE

   You can OWN a wooded or mountious property of your DREAMS
Think of it, your own private recreational hunting land LOADED
with deer, elk, wild turkey and lots of other game. To entertain
your friends, a mountain cabin, next to a stream built on many,
many acres for the cost of only ONE raffle ticket.

  We would like to mail you information, when our raffle starts in
January, just write or call us.
Thank you.

CHILDRENS RANCH RAFFLE
Rural Route 8. Box 238
PRARIE DU CHIEN, WISCONSIN 53821
1-800-600-0343 ext. 2104

46. Re: [[HFSIG:2835] newbie test and questions]

From: Alberto di Bene <dibene@usa.net>
Date: 28 Sep 98 15:09:39 MET DST
To: hfsig@tapr.org
Message-ID: <19980928130939.19254.qmail@www0a.netaddress.usa.net>
Rendered body type: text/plain

> This is my first entry to see if it works. If it does, I
> am recently retired from Intel and have some questions.
>
> 1. What is the status of Pentium (I&II) MMX digital applications
> software using nothing but a PC + sound board?

If you look around on Internet you will find quite a few of them.
<shameless-plug>
One of them is Hamview, which you can find at:
  http://www.freeyellow.com/members/padan
</shameless-plug>
To be honest, it doesn't take advantage of the MMX instructions, mainly to
make it compatible also with older Pentiums. It just perform all of the math
processing in the fast floating point unit of the chip. Maybe when MMX-2 will
be out, we will produce a version for it.

Alberto di Bene, I2PHD   dibene@usa.net

_____
Get free e-mail and a permanent address at http://www.netaddress.com/?N=1

47. Re: [HFSIG:2836] INCREASE SALES UP TO 200% - REFERRAL PROGRAM - $50-$150 PER REFERRAL

From: labay@zebra.net (Laurence Bay)
Date: Mon, 28 Sep 1998 10:51:05 -0500
To: <hfsig@tapr.org>
Message-ID: <19980928154849.AAA21973@default>
Rendered body type: text/plain

IF i get any more of these, I am going to take my name off the list.

-Larry
W1DOY

```
----------
> From: cdcyl@usa.net
> To: hfsig@tapr.org
> Subject: [HFSIG:2836] INCREASE SALES UP TO 200%  - REFERRAL PROGRAM -
$50-$150 PER REFERRAL
> Date: Monday, September 28, 1998 12:42 AM
>
>
>                         WIN
>
>         WISCONSIN OR OREGON WILDLIFE PARADISE
>
>    You can OWN a wooded or mountious property of your DREAMS
> Think of it, your own private recreational hunting land LOADED
> with deer, elk, wild turkey and lots of other game.  To entertain
> your friends, a mountain cabin, next to a  stream built on many,
> many acres for the cost of only ONE raffle ticket.
>
>   We would like to mail you information, when our raffle starts in
> January, just write or call us.
> Thank you.
>
>
> CHILDRENS RANCH RAFFLE
> Rural Route 8. Box 238
> PRARIE DU CHIEN, WISCONSIN 53821
> 1-800-600-0343 ext. 2104
>
```

48. Re: [HFSIG:2838] Re: INCREASE SALES UP TO 200% - REFERRAL PROGRAM - $50-$150 PER REFERRAL

```
From: W6RCecilA <w6rca@ibm.net>
Date: Mon, 28 Sep 1998 09:39:22 -0700
To: hfsig@tapr.org
Message-ID: <360FBBBA.4229@ibm.net>
Rendered body type: text/plain
```

Laurence Bay wrote:
>
> IF i get any more of these, I am going to take my name off the list.

Doesn't this imply that someone on the HFSIG mailing list is involved?
--
73, Cecil, W6RCA   http://people.delphi.com/CecilMoore

49. Re: [HFSIG:2837] Re: [newbie test and questions]

```
From: W6RCecilA <w6rca@ibm.net>
Date: Mon, 28 Sep 1998 20:46:30 -0700
To: hfsig@tapr.org
Message-ID: <36105816.94A@ibm.net>
Rendered body type: text/plain
```

Alberto di Bene wrote:
> If you look around on Internet you will find quite a few of them.
> <shameless-plug>

Thanks Alberto, does this software use DSP techinques or pulse-width
techniques?
--
73, Cecil, W6RCA   http://people.delphi.com/CecilMoore

50. Re: [[HFSIG:2840] Re: [newbie test and questions]]

```
From: Alberto di Bene <dibene@usa.net>
Date: 29 Sep 98 14:08:07 MET DST
To: hfsig@tapr.org
Message-ID: <19980929120807.10120.qmail@www0k.netaddress.usa.net>
Rendered body type: text/plain
```

> Thanks Alberto, does this software use DSP techinques or pulse-width
> techniques?
> --
> 73, Cecil, W6RCA   http://people.delphi.com/CecilMoore

Cecil, if you are referring to Hamview, it uses FFT to compute the spectral
display and to implement the FIR filtering, and a (marginally stable) IIR
biquad section for the CW filtering. Optionally the spectral display can be
computed using the Blackmann-Tukey estimator, via a mouse-selectable option on

the main panel.

73, Alberto, I2PHD   dibene@usa.net

_____
Get free e-mail and a permanent address at http://www.netaddress.com/?N=1

October 1998

Original source file: hfsig.9810
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9810

1. APRS Reed Soloman coding HF?

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Thu, 1 Oct 1998 12:51:35 -0400 (EDT)
To: hfsig@tapr.org
Cc: aprssig@tapr.org
Message-ID: <Pine.GSO.4.04.9810011237370.15574-100000@arctic>
Rendered body type: text/plain

APRS uses 300 baud PACKET on HF, which is about the worst mode. But it works. BUT IT COULD WORK BETTER. The first change is the new compression format for 1999 which will reduce the size of the position from 27 down to 13 bytes. Thus a much shorter packet. Thus MUCH better success.

But here is a new idea:

Lets experiment with APRS HF packets with PASSALL-ON so that at least we get a copy of all packets even those with errors. Then we add some additional "coding" bits to be able to do error correction. Is there anyone that would like to look at this as a means of GREATLY improving mobile position reporting on HF?

The HF position format with the usual packet header looks like this:

```
WB4APR>APRS:tYYYYXXXX$csT...
```

All of the above 13 bytes use the 91 printable asccii characters, so most redundancy has already been removed. I want to now take that data, and do some "Reed-Soloman" or some such coding to get the same information through a receiving TNC with PASSALL ON so that its errors can be corrected and thus get much higher success with no HARDWARE changes...

```
WB4APR>APRS:??????????????????
```

Is there anyone out there that wants to take this challenge?
Who can work up the coding and decoding algorithm for me (that even I can understand).... :-)

Bob

2. Re: [HFSIG:2842] APRS Reed Soloman coding HF?

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Thu, 1 Oct 1998 12:21:35 -0500 (CDT)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.981001121531.911I-100000@www.kelly-afb.org>
Rendered body type: text/plain

Bob,

Way back before Desert Storm, I was asked to outfit an Air Force Civil Engineering group with HF packet so they could send binary files via HF from the Caribean back to Kelly AFB.

They had a military HF transceiver made by Harris RF Comm and we got them a PK-232. They learned how to use the equipment and software and headede for the Caribean.

They started out at 300 baud packet with all the parameters optimized for HF operation. Their thruput was terrible. Since I had forgotten all about the research that SRI had done on RTTY (RATT) just after WWII I was really scratching my head. Then some folks from Harris RF Comm reminded me that the reason that RATT stopped at 110 baud was that above 150 baud you had propagation problems.

So, we reduced the packet aud rate to 110 baud and the thruput jumped up to almost no re-tries where at 300 baud we had constant re-tries.

Try 110 baud packet...it really works much better than 300 baud.

Walt

```
    ========================================================================
    |           In a world without fences, who needs Gates?                |
    |----------------------------------------------------------------------|
```

```
|        If I wanted "Windows", I'd knock a hole in my wall.          |
=======================================================================
|                                  |                                  |
|                                  | The Micro$oft operating system   |
| Walt DuBose - K5YFW              | didn't get as bad as it is       |
| E-Mail k5yfw@www.kelly-afb.org   | overnight.  I has taken over 10  |
| Business Telephone: (210)925-1171| years of careful, calculated,    |
|     Home Telephone: (210)696-3196| development.                     |
|                                  |                                  |
=======================================================================
|   Using MS is like eating high class dog food...its still dog food. |
=======================================================================
```

On Thu, 1 Oct 1998, Bob Bruninga wrote:

> APRS uses 300 baud PACKET on HF, which is about the worst mode. But it
> works. BUT IT COULD WORK BETTER. The first change is the new compression
> format for 1999 which will reduce the size of the position from 27 down to
> 13 bytes. Thus a much shorter packet. Thus MUCH better success.
>
> But here is a new idea:
>
> Lets experiment with APRS HF packets with PASSALL-ON so that at least we
> get a copy of all packets even those with errors. Then we add some
> additional "coding" bits to be able to do error correction. Is there
> anyone that would like to look at this as a means of GREATLY improving
> mobile position reporting on HF?
>
> The HF position format with the usual packet header looks like this:
>
>   WB4APR>APRS:tYYYYXXXX$csT...
>
> All of the above 13 bytes use the 91 printable asccii characters, so most
> redundancy has already been removed. I want to now take that data, and do
> some "Reed-Soloman" or some such coding to get the same information
> through a receiving TNC with PASSALL ON so that its errors can be
> corrected and thus get much higher success with no HARDWARE changes...
>
>   WB4APR>APRS:??????????????????
>
> Is there anyone out there that wants to take this challenge?
> Who can work up the coding and decoding algorithm for me (that
> even I can understand).... :-)
>
> Bob
>
>
>

3. Re: [HFSIG:2842] APRS Reed Soloman coding HF?

From: Doug Faunt N6TQS +1-510-655-8604 <faunt@netcom.com>
Date: Thu, 1 Oct 1998 10:36:09 -0700 (PDT)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <199810011736.KAA24612@netcom6.netcom.com>
Rendered body type: text/plain

Date: Thu, 1 Oct 1998 11:56:41 -0500 (CDT)
   From: Bob Bruninga <bruninga@nadn.navy.mil>

   APRS uses 300 baud PACKET on HF, which is about the worst mode. But it
   works. BUT IT COULD WORK BETTER. The first change is the new compression
   format for 1999 which will reduce the size of the position from 27 down to
   13 bytes. Thus a much shorter packet. Thus MUCH better success.

What about doing away with the TNC altogether, using some HF modem or
TU type unit (which can be very small and cheap), and doing all the
decoding in the APRS software?

73, doug

4. RE: [HFSIG:2842] APRS Reed Soloman coding HF?

From: "Lanier, Robert" <rlanier@harris.com>
Date: Thu, 1 Oct 1998 13:37:38 -0400
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <275399FB18C4D111871300805FBEB72F01215BB5@corpmx6.ess.harris.com>
Rendered body type: text/plain

well, I know Phil Karn has some Reed-Soloman coding stuff on his web page.

Tony

-----Original Message-----
From: hfsig@tapr.org [SMTP:hfsig@tapr.org]
Sent: Thursday, October 01, 1998 11:57 AM
To: hfsig@tapr.org
Subject: [HFSIG:2842] APRS Reed Soloman coding HF?

APRS uses 300 baud PACKET on HF, which is about the worst mode. But it works. BUT IT COULD WORK BETTER. The first change is the new compression format for 1999 which will reduce the size of the position from 27 down to 13 bytes. Thus a much shorter packet. Thus MUCH better success.

But here is a new idea:

Lets experiment with APRS HF packets with PASSALL-ON so that at least we get a copy of all packets even those with errors. Then we add some additional "coding" bits to be able to do error correction. Is there anyone that would like to look at this as a means of GREATLY improving mobile position reporting on HF?

The HF position format with the usual packet header looks like this:

WB4APR>APRS:tYYYYXXXX$csT...

All of the above 13 bytes use the 91 printable asccii characters, so most redundancy has already been removed. I want to now take that data, and do some "Reed-Soloman" or some such coding to get the same information through a receiving TNC with PASSALL ON so that its errors can be corrected and thus get much higher success with no HARDWARE changes...

WB4APR>APRS:??????????????????

Is there anyone out there that wants to take this challenge?
Who can work up the coding and decoding algorithm for me (that even I can understand).... :-)

Bob

5. Re: [HFSIG:2844] Re: APRS Reed Soloman coding HF?

From: malberts@ncs.com.na (Mike Alberts)
Date: Mon, 05 Oct 1998 09:31:48 +0200
To: hfsig@tapr.org
Message-ID: <361875E4.D7F4407@ncs.com.na>
Rendered body type: text/plain

Hi ,
I Need more help .....
I want to put up a packet gateway from NAMIBIA V51 land but need equipment , in the form of TNC's and Antenna's.
Where could I mail a message on the net to HAMS who willbe willing to get Namibia on the Packet gateway.
Thank's
73's
Mike
V51MA

Doug Faunt N6TQS +1-510-655-8604 wrote:

> Date: Thu, 1 Oct 1998 11:56:41 -0500 (CDT)
> From: Bob Bruninga <bruninga@nadn.navy.mil>
>
> APRS uses 300 baud PACKET on HF, which is about the worst mode. But it
> works. BUT IT COULD WORK BETTER. The first change is the new compression
> format for 1999 which will reduce the size of the position from 27 down to
> 13 bytes. Thus a much shorter packet. Thus MUCH better success.
>
> What about doing away with the TNC altogether, using some HF modem or
> TU type unit (which can be very small and cheap), and doing all the
> decoding in the APRS software?
>
> 73, doug

6. email gateways on HF

From: Bill Bennett N7DZ <WizardOfAz@Cyberdude.com>
Date: Mon, 05 Oct 1998 09:27:17 -0700
To: hfsig@tapr.org
Message-ID: <3618F365.AE5F40AF@Cyberdude.com>
Rendered body type: text/plain

Can anyone provide info or pointers to info about gateways between HF
digital and internet email? One great use of such a link would be to
send/receive email when camping in a location not near enough to any VHF
packet node.

Thanks for any feedback.

Bill - N7DZ

---

7. Re: [HFSIG:2847] email gateways on HF

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Mon, 5 Oct 1998 12:49:59 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.4.04.9810051244590.21711-100000@arctic>
Rendered body type: text/plain

On Mon, 5 Oct 1998, Bill Bennett N7DZ wrote:

> Can anyone provide info or pointers to info about gateways between HF
> digital and internet email? One great use of such a link would be to
> send/receive email when camping...

APRS has now grown to be a worldwide message capabiility. Every APRS
packet whether VHF, UHF, HF, or satellite is all monotired and fed into
199.227.86.221 port 23 by all the IGates and consequently these are
delivered to all other IGAtes (with all dupes removed). Thus, any packet
launched anwhere where there is APRS infrastructure, can bb delivered
anywhere.

This is ALL transparent to the user! If he sends an APRS message to
station X anywhere in the world, and an Igate near station X sees the
mesage on the APRSserv stream, it will pull it off and send it back to RF.
Thus, real-time messages anwyhere from anywhere.

Now to this we are adding EMAIL engines. THat can look for messages
addressed to EMAIL. THese will be pulled off the APRServe stream and send
as EMAIL to the receipient!

with HF, this system is the the worlds largest HF DIversity Reception
system in the world. Dozens of HF APR packet stations are monitoring the
10.151 LSB frequency and if ANY ONE Of them hears a packet, then it is
shipped into APRSSERVE for everyone else...

bob

---

8. Re: [HFSIG:2847] email gateways on HF

From: "Frank Mondt" <n0sms@janics.com>
Date: Mon, 5 Oct 1998 17:33:40 -5
To: hfsig@tapr.org
Message-ID: <199810052233.RAA27093@tapr.org>
Rendered body type: text/plain

Date sent: Mon, 5 Oct 1998 11:27:40 -0500 (CDT)
Send reply to: hfsig@tapr.org
From: Bill Bennett N7DZ <WizardOfAz@Cyberdude.com>
To: hfsig@tapr.org
Subject: [HFSIG:2847] email gateways on HF

> Can anyone provide info or pointers to info about gateways between HF
> digital and internet email? One great use of such a link would be to
> send/receive email when camping in a location not near enough to any VHF
> packet node.
>
> Thanks for any feedback.
>
> Bill - N7DZ
>

Hi Bill, It's funny that you should ask. Here is the release information for a program
that can do just that if used with an FBB BBS with an HF port. I am an ex FBB sysop
and am wondering if this might not give me an excuse to reopen my BBS Hi.

From: "The Brown Family" <acts29@missions4Him.com>
To: "FBB Mailing List" <fbb@f6fbb.org>
Subject: BBSPop Program
Date sent: Mon, 5 Oct 1998 08:45:38 -0500

Sysops,

The BBSPop program is ready for Beta test. This is a pop3/smtp program
and server for FBB. You can read information on it at
http://www.Missions4Him.com/bbspop

If you do not have Net access then send e-mail to me
RadioMail@Missions4Him.com
put BBSPop Info in the subject and I will forward information to you
about
the program

Kenton, Saundi, and Josiah Brown
Missionaries to Honduras
acts29@Misssions4Him.com
http://www.Missions4Him.com
ICQ: 795839

All these promises are made to us, my dear friends. So then, let us
purify ourselves from everything that makes body or soul unclean, and
let
us be completely holy by living in awe of God. 2 Corinthians 7:1 (Good
News New Testament)

Shareware Online Bible for Windows: http://www.onlinebible.org

>

Frank & Shirley

n0sms@janics.com <---NOTE!! EMAIL ADDRESS CHANGE!!

9. Re: [HFSIG:2848] Re: email gateways on HF

From: John Dolan <jdolan@erols.com>
Date: Wed, 07 Oct 1998 11:51:04 -0700
To: hfsig@tapr.org
Message-ID: <361BB818.4B19@erols.com>
Rendered body type: text/plain

>> Can anyone provide info or pointers to info about gateways between HF
>> digital and internet email? One great use of such a link would be to
>> send/receive email when camping...

> APRS has now grown to be a worldwide message capabiility. Every APRS
> packet whether VHF, UHF, HF, or satellite is all monotired and fed into
> 199.227.86.221 port 23 by all the IGates and consequently these are
> delivered to all other IGAtes (with all dupes removed). Thus, any
> packet launched anwhere where there is APRS infrastructure, can be
> delivered anywhere.

I realize this is true... and like many new concepts, one doesn't always
immediately think about all the ways of using APRS. For those of us
just learning... it is limited in this case to 50 or so characters per
message; am I right, Bob?
--

Regards,
John Dolan
K3FIG
jdolan@erols.com

10. Re: [HFSIG:2850] Re: email gateways on HF

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Wed, 7 Oct 1998 14:01:20 -0400 (EDT)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.4.04.9810071359520.10799-100000@arctic>
Rendered body type: text/plain

On Wed, 7 Oct 1998, John Dolan wrote:

> > APRS has now grown to be a worldwide message capabiility. Every APRS
> > packet whether VHF, UHF, HF, or satellite is all monotired and fed into
> > 199.227.86.221 port 23 by all the IGates and consequently these are
> > delivered to all other IGAtes (with all dupes removed). Thus, any
> > packet launched anwhere where there is APRS infrastructure, can be
> > delivered anywhere.
>
> I realize this is true... and like many new concepts, one doesn't always
> immediately think about all the ways of using APRS.   For those of us
> just learning... it is limited in this case to 50 or so characters per
> message; am I right, Bob?

It is normally 63 or 67 (I forget) but if you want to be able to send and
receive from the new Kenwood HT (with built in TNC and display), then 45
characters is the limit. But you can send multiple lines if needed...

bob

11. Re: [HFSIG:2837] Re: [newbie test and questions]

From: W6RCecilA <w6rca@ibm.net>
Date: Thu, 08 Oct 1998 17:58:39 -0700
To: hfsig@tapr.org
Cc: ke4zv@bellsouth.net
Message-ID: <361D5FBF.C55@ibm.net>
Rendered body type: text/plain

Who is doing HF digital development work with the DSP56002EVM
board? I just ordered one. Anybody done AMTOR or PACTOR with
that board?

Is there a standard PC sound board that is used for this kind
of development or a standard set of 16-bit sound board drivers
that developers use with a PC? I have the drivers for a PAS-16
but I'm sure they don't work with any other sound cards.
--
73, Cecil, W6RCA   http://people.delphi.com/CecilMoore

12. Re: [HFSIG:2852] Re: [newbie test and questions]

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Fri, 09 Oct 1998 23:06:48 +0200
To: hfsig@tapr.org
Message-ID: <361E7AE8.2988C25C@ife.ee.ethz.ch>
Rendered body type: text/plain

W6RCecilA wrote:

> Is there a standard PC sound board that is used for this kind
> of development or a standard set of 16-bit sound board drivers
> that developers use with a PC? I have the drivers for a PAS-16
> but I'm sure they don't work with any other sound cards.

Any card whose driver is low latency enough will do.
Full duplex capable cards are easier, as they do not need
to switch between receive and transmit. Some cards may do lengthy
calibration when changing the direction (which can be turned off,
but the driver needs to do it).

Tom

13. Software for DSP56002EVM.

From: "M. Greenman" <coombedn@ihug.co.nz>
Date: Sat, 10 Oct 1998 20:38:34 +1300
To: <w6rca@ibm.net>
Cc: <hfsig@tapr.org>
Message-ID: <199810100739.UAA26693@smtp2.ihug.co.nz>
Rendered body type: text/plain

Cecil,
There is quite a lot of software for the EVM. I suggest you look at
software by SP9VRC, KC7WW (available on TAPR), and G3PLX. There is
AMTOR/PACTOR software by KC7WW, and also PSK31, MT63, Feld Hell and
MT-Hell.

Have a look on 14063.5 LSB between 0600z and 0800z, and again at around
1900z. You are certain to hear EVMs on the air.

If you cannot find them, I can give you some sites to look at, but start

with TAPR, as they have a whole page dedicated to the EVM.

There is sound card software for RTTY, and Rob ZL2AKM is working on a
version for PACTOR. G3PPT has excellent Feld-Hell and MT-Hell software for
sound card. That and the G3PLX software is on my site, at:

members.xoom.com/ZL1BPU/Contents.html

73,

Murray ZL1BPU

(I too have an EVM - and would love to work you on MT-Hell!)

14. Re: [HFSIG:2852] Re: [newbie test and questions]

From: Eduardo Jacob <edu@kender.es>
Date: Mon, 12 Oct 1998 19:55:35 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19981012195535.00899ae0@kender.es>
Rendered body type: text/plain

At 20:03 08/10/98 -0500, you wrote:
>Who is doing HF digital development work with the DSP56002EVM
>board? I just ordered one. Anybody done AMTOR or PACTOR with
>that board?

Perhaps you've already been there, but you can take a look at my home
page, I have evrything I have find for the EVM on it.

http://det.bi.ehu.es/~jtpjatae/ham.html

Regards

Eduardo/EA2BAJ

15. New version of PSK31 for the EasyDSP

From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
Date: Thu, 15 Oct 1998 13:16:52 +0200
To: psk31@bipt106.bi.ehu.es
Message-ID: <3.0.5.32.19981015131652.008693e0@bicc00.bi.ehu.es>
Rendered body type: text/plain

I have just received a version of PSK31evm301 (just to remind you, the
latest version of PSK31 for the EVM on DOS) for the nice Motorola 56002
based EasyDSP.

It's available here:

ftp://det.bi.ehu.es/pub/ham/psk31/psk31eas301.zip

Just to remind you, the psk31 homepage is at

http://aintel.bi.ehu.es/psk31.html

I don't have an EasyDSP unit, so I won't be able to solve you any
problems. You can take a look at it at

http://ourworld.compuserve.com/homepages/HFFAX/toc32.htm

Regards

Eduardo/EA2BAJ

Eduardo EA2BAJ
DSP and HAM: http://det.bi.ehu.es/~jtpjatae/ham.html
VHF PACKET: EA2BAJ @ EA2URV
PGP Key avlb: Send email w/subject: PGPKEY
fingerprint : D8 FA 54 49 F7 40 BA AA 09 1A 73 40 A4 26 ED BD

16. Re: [HFSIG:2856] New version of PSK31 for the EasyDSP

From: W6RCecilA <w6rca@ibm.net>
Date: Sat, 17 Oct 1998 20:34:04 -0700
To: hfsig@tapr.org
Message-ID: <362961AC.607@ibm.net>
Rendered body type: text/plain

Eduardo Jacob wrote:
>

>   I have just received a version of PSK31evm301 (just to remind you, the
> latest version of PSK31 for the EVM on DOS) for the nice Motorola 56002
> based EasyDSP.

Thanks Eduardo, where can I find the specification for PSK31? Thanks -
--
73, Cecil, W6RCA   http://people.delphi.com/CecilMoore

17. Re: [HFSIG:2857] Re: New version of PSK31 for the EasyDSP

From: Eduardo Jacob <edu@kender.es>
Date: Sun, 18 Oct 1998 08:52:54 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19981018085254.00821100@kender.es>
Rendered body type: text/plain

>Thanks Eduardo, where can I find the specification for PSK31? Thanks -

        It's here. Although you should get the windows package to get the 255
characters explanation.

        http://aintel.bi.ehu.es/psk31theory.html

        In any case, Peter is always willing to explain details of PSK31

        Eduardo/EA2BAJ

18. TAPR PSK Modem?

From: "Fernando M. I. Carreiro" <fmi@carreiro.com>
Date: Thu, 22 Oct 1998 00:52:09 +0100
To: <dsp@tapr.org>, <tapr-tnc@tapr.org>, <hfsig@tapr.org>
Message-ID: <005501bdfd4e$0dc48020$e6ffa8c0@pc001>
Rendered body type: text/plain

I am looking for information and technical specifications on the old TAPR PSK Modem. Can anyone perhaps tell me, from where I can download such information. I am particularly interested in the "costas loop" used for carrier recovery.
    Thank you and regards,

Fernando M. I. Carreiro
mailto:fmi@carreiro.com
http://www.fmi.carreiro.com
callsign:ZS6BLF

19. TAPR PSK Modem? (HTML Removed)

From: "Fernando M. I. Carreiro" <fmi@carreiro.com>
Date: Thu, 22 Oct 1998 16:05:57 +0100
To: <dsp@tapr.org>, <tapr-tnc@tapr.org>, <hfsig@tapr.org>
Message-ID: <003801bdfdd0$2e90ac20$e6ffa8c0@pc001>
Rendered body type: text/plain

I am looking for information and technical specifications on the old TAPR PSK Modem. Can anyone perhaps tell me, from where I can download such information. I am particularly interested in the "costas loop" used for carrier recovery.
    Thank you and regards,

Fernando M. I. Carreiro
mailto:fmi@carreiro.com
http://www.fmi.carreiro.com
callsign:ZS6BLF

20. DWM ?

From: Danie Brynard <BRYD@kidd.Co.Za>
Date: Thu, 22 Oct 1998 13:47:00 +0200
To: hfsig@tapr.org
Message-ID: <s62f375d.095@kidd.Co.Za>
Rendered body type: text/plain

Can anybody confirm the amazing parameters that IAS is quoting for their DWM (digital wave modulation) system ? Apparently violates Shannon's Law. Be nice for HF comms..

Danie

PS claim is they can transmit 1000 times more in the same bandwidth

21. Re: [HFSIG:2861] DWM ?

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Thu, 22 Oct 1998 19:33:59 +0100
To: hfsig@tapr.org
Message-ID: <362F7A97.3D0D@marconicomms.com>
Rendered body type: text/plain

Danie Brynard wrote:
>
> Can anybody confirm the amazing parameters that IAS is quoting for their
> DWM (digital wave modulation) system ? Apparently violates Shannon's
> Law. Be nice for HF comms..
>
> Danie
>
> PS claim is they can transmit 1000 times more in the same bandwidth

How about a URL?

--
Suffering from `changed e-mail address syndrome'

A.N.Martin E-mail anthony.n.martin@marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK

22. Re: [HFSIG:2861] DWM ?

From: "Mike Fahmie" <wa6zty@hotmail.com>
Date: Thu, 22 Oct 1998 11:52:39 PDT
To: hfsig@tapr.org
Message-ID: <19981022185240.6262.qmail@hotmail.com>
Rendered body type: text/plain

At 12:43 PM 10/22/98 -0500, you wrote:
>Can anybody confirm the amazing parameters that
>IAS is quoting for their DWM (digital wave
>modulation) system ? Apparently violates Shannon's
>Law. Be nice for HF comms..
>
>Danie
>
>PS claim is they can transmit 1000 times more
>in the same bandwidth
>

According to an IAS press release dated Sept 25, 1998, the Securities
and Exchange Commission has some questions as well.

-Mike-
WA6ZTY

_____
Get Your Private, Free Email at http://www.hotmail.com

23. Re: DWM ? (URL Info)

From: "Fernando M. I. Carreiro" <fmi@carreiro.com>
Date: Thu, 22 Oct 1998 21:13:27 +0100
To: <hfsig@tapr.org>
Message-ID: <009001bdfdf8$701b5c80$e6ffa8c0@pc001>
Rendered body type: text/plain

I searched the Internet and found this URL about the info below:
http://www.iaus.com/news.htm

Fernando M. I. Carreiro
mailto:fmi@carreiro.com
http://www.fmi.carreiro.com
callsign:ZS6BLF

-----Original Message-----
From: Danie Brynard <BRYD@kidd.co.za>
To: hfsig@tapr.org <hfsig@tapr.org>
Date: Thursday, 22 October, 1998 19:20
Subject: [HFSIG:2861] DWM ?

>Can anybody confirm the amazing parameters that IAS is quoting for their
>DWM (digital wave modulation) system ? Apparently violates Shannon's
>Law. Be nice for HF comms..
>
>Danie
>
>PS claim is they can transmit 1000 times more in the same bandwidth
>

24. Why so little participation in USA?

From: forrerj@peak.org (Johan Forrer)
Date: Sat, 24 Oct 1998 09:27:07 -0800
To: HFSIG@TAPR.ORG
Message-ID: <199810241630.JAA17587@peak.org>
Rendered body type: text/plain

Here is my pea:

Are there still experimentors out there that are interested in the newer
crop of digital modes for HF? Meaning actually getting on the air and
playing with it -- helping with development work instead of just talking
about it? What can be done to encourage participation?

The bands are good right now -- equipment requirements are
quite modest. For example; I run 40W output into a vertical antenna
and using MTHELL or MT63 and the 56002EVM, can work
VK2DSG with excellent results and regularity. These two modes are
not critical as far as frequency accuracy is concerned. If you are
within +/- 5 Hz, that is good. Most transceivers can do that.

There also is PSK31, which now is avalailable for several platforms:
56002EVM, TI320C50, SHARC EZ-KIT, and even Linux
(using a sound card). This one works very well, even under poor
conditions. It does, however require very very precise frequency
tolerances, i.e., 2 Hz is about the limit. A rig like the ICOM 706 can
do this quite easily, so that is no big drawback.

As an alternative, VE2IQ's C-BPSK system also is a viable
alternative. It has a somewhat-larger support group here in the USA,
and regular trans-Atlantic contacts are commonly taking place. As
for PSK31, frequency accuracy is most important. Equipment
requirements otherwise being quite modest.

But there also is the 15-tone AX.25 modem that will do TCP/IP over
HF. It runs at a respectable 3800 bps (with error control 800 bps). I
personally can attest to how well it works from channel simulator tests
and over-the-air testing against worthy competitors like for example
PACTOR II and CLOVER-2000. This modem, presently only runs
on the 56002EVM. It is designed for application in a HF network
topology and some kind of terminal program or NOS (KA9Q,
JNOS, Linux, or equivalent) is needed to use or integrate it into
existing networks. Knowledge of HF digital communications and
computer networks are useful before trying this.

These are just a couple of worthy pursuits --- Does anyone have
any idea why there is so little participation here in the USA?

Is it due to a lack of basic education, i.e.,
"How-To" information ....
possibly too little being published about it? ...
Difficulty in getting the hardware together? ...
Where to get the software? ...

What can we do to attract more participation?

73

Johan, KC7WW

P.S.
For brevity, I did not even mention all the new digital stuff that
is out there. I sincerely aplogise for that.

25. Re: [HFSIG:2865] Why so little participation in USA?

From: "Paulus" <paulstra@xs4all.nl>
Date: Sat, 24 Oct 1998 19:12:29 +0200
To: <hfsig@tapr.org>
Message-ID: <000901bdff71$7bbcccc0$ee816dc2@machientje>

Rendered body type: text/plain

Hello all,
I quite agree with Johan KC7WW because in Europe the activity is also an bit low, but there are stations !, not many though, but just enough to keep PSK31, Coherent BPSK and MThell/Feldhell going.
Mt63 and 15 tones QPSK not often though hi.
G3iRM published in Datacom and CQ-DL about coherent-BPSK and I wrote an article in the Dutch CQ-PA magizine about all those fine new digital modes and platforms to run it. We did get some response, and we met allready an few new C-BPSK stations.
Off course we will go on "advertising".
Perhaps an publication in QST, about how (easy) and without making an big investement, you can run all these "ham" modes, will do the job.

BTW: We just at 1800-1900 (24 october) had an QSO on 14.081, 18.081 and 21.081 with G3GKI and AA1A.
I run an very modest station, 2 homemade Indoor Magnetic loops ! and about 30 watts from an IC-706 MK2 with TCXO.

So I am Qrv to test an help.

73 de Paul, PA0OCD
paulstra@xs4all.nl

26. Fw: [BPSK] todays sigs

From: "Paulus" <paulstra@xs4all.nl>
Date: Sat, 24 Oct 1998 19:19:58 +0200
To: <hfsig@tapr.org>
Cc: "Dave" <daveaa1a@xensei.com>
Message-ID: <003101bdff72$873e3ec0$ee816dc2@machientje>
Rendered body type: text/plain

-----Oorspronkelijk bericht-----
Van: aa1a@radiocom.net <aa1a@radiocom.net>
Aan: bpsk@qth.net <bpsk@qth.net>; lowfer@qth.net <lowfer@qth.net>
Datum: zaterdag 24 oktober 1998 19:14
Onderwerp: [BPSK] todays sigs

>
>Good copy in weak sigs today on 14081 @ 16z from G3GKI PA0OCD G3FMW.
>
>Last 2 days we have qsy'd up and made it to 21 mhz but nil on 10m.
>
>Last night copy on 'SOLAR ' beacon @ 1690.590 ms50 et1 caused 514
>complete hits for a 12 hour file and over a thousand partial hits.
>Distance is about 350 miles and covering sigs were S9 from a station
>down the coast.
>
>73s from Dave
>
>==========
=
> Dave Riley - AA1A - www.radiocom.net - the light years are getting shorter
>==========
=
>
>-----
>Submissions: bpsk@qth.net
>Subscription/removal: bpsk-request@qth.net
>

27. Re: [HFSIG:2865] Why so little participation in USA?

From: labay@zebra.net (Laurence Bay)
Date: Sat, 24 Oct 1998 12:15:56 -0500
To: <hfsig@tapr.org>
Message-ID: <000201bdff90$740378c0$7aee0cd1@default>
Rendered body type: text/plain

-----Original Message-----
From: Johan Forrer <forrerj@peak.org>
To: hfsig@tapr.org <hfsig@tapr.org>
Date: Saturday, October 24, 1998 12:02 PM
Subject: [HFSIG:2865] Why so little participation in USA?

>Here is my pea:
>

>Are there still experimentors out there that are interested in the newer
>crop of digital modes for HF? Meaning actually getting on the air and
>playing with it -- helping with development work instead of just talking
>about it? What can be done to encourage participation?
>
>The bands are good right now -- equipment requirements are
>quite modest. For example; I run 40W output into a vertical antenna
>and using MTHELL or MT63 and the 56002EVM, can work
>VK2DSG with excellent results and regularity. These two modes are
>not critical as far as frequency accuracy is concerned. If you are
>within +/- 5 Hz, that is good. Most transceivers can do that.
>
>There also is PSK31, which now is avalailable for several platforms:
>56002EVM, TI320C50, SHARC EZ-KIT, and even Linux
>(using a sound card). This one works very well, even under poor
>conditions. It does, however require very very precise frequency
>tolerances, i.e., 2 Hz is about the limit. A rig like the ICOM 706 can
>do this quite easily, so that is no big drawback.
>
>As an alternative, VE2IQ's C-BPSK system also is a viable
>alternative. It has a somewhat-larger support group here in the USA,
>and regular trans-Atlantic contacts are commonly taking place. As
>for PSK31, frequency accuracy is most important. Equipment
>requirements otherwise being quite modest.
>
>But there also is the 15-tone AX.25 modem that will do TCP/IP over
>HF. It runs at a respectable 3800 bps (with error control 800 bps). I
>personally can attest to how well it works from channel simulator tests
>and over-the-air testing against worthy competitors like for example
>PACTOR II and CLOVER-2000. This modem, presently only runs
>on the 56002EVM. It is designed for application in a HF network
>topology and some kind of terminal program or NOS (KA9Q,
>JNOS, Linux, or equivalent) is needed to use or integrate it into
>existing networks. Knowledge of HF digital communications and
>computer networks are useful before trying this.
>
>These are just a couple of worthy pursuits --- Does anyone have
>any idea why there is so little participation here in the USA?
>
>Is it due to a lack of basic education, i.e.,
>"How-To" information ....
>possibly too little being published about it? ...
>Difficulty in getting the hardware together? ...
>Where to get the software? ...
>
>What can we do to attract more participation?
>
>73
>
>Johan, KC7WW
>
>P.S.
>For brevity, I did not even mention all the new digital stuff that
>is out there. I sincerely aplogise for that.
>

No need to apologize, Johan.

But you just mentioned in passing, some of the more garden variety modes
like Pactor, Amtor. I am just getting into these and with 5 yr-old DOS
software, I hate to spend the bucks before I know how good what I am buying
will be.

I haven't been able to get into Pactor much because the s/w I am using is
poorly documented and I lose contacts.

I don't hear much activity on RTTY either, which is discouraging.

I am using Kam Plus, TS450-SAT and dipoles on all bands. A modest beginning.

Appreciated hearing your comments.

-Larry
W1DOY

Mobile, AL

28. Re: [HFSIG:2865] Why so little participation in USA?

From: Jim Sanford <jsanfor@exis.net>
Date: Sat, 24 Oct 1998 19:09:37 -0400
To: hfsig@tapr.org
Message-ID: <36325E30.9C7D1854@exis.net>
Rendered body type: text/plain

Johan Forrer wrote:

> Here is my pea:
>
> Are there still experimentors out there that are interested in the newer
> crop of digital modes for HF? Meaning actually getting on the air and
> playing with it -- helping with development work instead of just talking
> about it? What can be done to encourage participation?
>

Absolutely! Unfortunately, in my case, job permits little time for such
time consuming advanced work -- at the moment.

> These are just a couple of worthy pursuits --- Does anyone have
> any idea why there is so little participation here in the USA?
>
> Is it due to a lack of basic education, i.e.,
> "How-To" information ....
>

I think this is part of it.

> possibly too little being published about it? ...

Agree

>
> Difficulty in getting the hardware together? ...
>

Absolutely

> Where to get the software? ...
>

another reason.

>
> What can we do to attract more participation?
>

Make it easier. One thing I've seen with satellite work, is that most hams
want something pretty close to plug and play. Satellite work is not, hence
relatively small percentage of aficionados. Same for advanced digital
modes, especially if purchase of something special like an EVM or software
is required. If mods to rig are required, that also cuts out many.

Frankly, I like the best of both worlds -- the convenience of using the
installed sound card, but I also like the stand-alone EVM etc.,.....seems a
waste to tie up all the power in a 200 mhz pentium mmx just to talk
digitally. So, suggest all work possible be designed from the start to go
either way, depending on user's choices.

I follow carefully your work, and most of what you write (and links to the
software) get saved for the day I have a few minutes to myself to imbibe in
the dsp world. Keep up the great work.

73, Jim
wb4gcs@amsat.org

29. Re: [HFSIG:2869] Re: Why so little participation in USA?

From: W6RCecilA <w6rca@ibm.net>
Date: Sat, 24 Oct 1998 17:29:27 -0700
To: hfsig@tapr.org
Message-ID: <363270E7.758@ibm.net>
Rendered body type: text/plain

Jim Sanford wrote:
> Make it easier.
> Frankly, I like the best of both worlds -- the convenience of using the
> installed sound card, but I also like the stand-alone EVM etc.,.....seems a

> waste to tie up all the power in a 200 mhz pentium mmx just to talk
> digitally.

I just retired from Intel and am far behind you guys so this is just an opinion. I think what we need is a mode that runs 100% on a PC/sound card with neglible external electronics required besides the transceiver. The mode could have selectable levels of complexity and throughput depending on conditions. I'm talking about something that on the lowest end runs about ten baud ASCII - roughly equivalent to 12wpm CW. Such a mode should be able to best a human CW operator, something I've always wanted to do.

I assume that most PCs nowadays are SB-16 compatible. Are there any free application software packages available for assembly language, BASIC, or 'C', or object code suitable for linking, or DOS calls?

I just ordered a DSP60002EVM eval board and the TAPR box to go with it. I have a Pentium MMX 166 with a Yamaha chip set. I'm getting ready to do some development probably starting with something like low baud rate ASCII/FSK with some error correction.
--
73, Cecil, W6RCA http://people.delphi.com/CecilMoore

30. Coding i HF data transmission

From: Halvor Liland <Halvor.Liland@bo.online.no>
Date: Sun, 25 Oct 1998 12:39:16 +0100 (MET)
To: hfsig@tapr.org
Message-ID: <199810251139.MAA28048@online.no>
Rendered body type: text/plain

I am working with a study in wireless digital transmission. I have read som texts that confuse me about what is the best coding, convolutional codes or block codes like Reed-Solomon, for HF data transmission. Is there anybody in this SIG that have experience or some teoretical reasons for the best coding type.

LA7BO Halvor

31. Re: [HFSIG:2871] Coding i HF data transmission

From: forrerj@peak.org (Johan Forrer)
Date: Sun, 25 Oct 1998 09:44:48 -0800
To: hfsig@tapr.org
Message-ID: <199810251748.JAA29678@peak.org>
Rendered body type: text/plain

Halvor,

Just a few thoughts ....

A lot have been said about this topic in the past on this list. Of course as it relates to HF digital communications. It may be a good idea to search the archive (accessible from the TAPR web site) with a subject of "coding".

One should think about a coding system that is taylored for a specific "channel" as something that works for HF probably will not necessarily work well on UHF. The coding has to do it work on different kinds of channel pertubations, some unique for the that channel.

The main and major factor on HF channels is destructive multipath and fading affects. Second to that is noise behaviour, particularly a kind that may be modeled by Markovian statistics rather than the usual Gaussian model. The Watterson ionosperic propagation model is gereally accepted as a good one.

With that bit of background, you want to design the coding system to effectively deal with signals that has randomly distorted bit edges, against a burst-noise background. What seems to work well are the use time- and frequency diversity techniques. Both bock and convolutional codes have their place, often in combination, i.e., a convolutional inner code wrapped inside an appropiate block code. Time-diversity (interleaving) methods does wonderful things for HF, so does frequency diversity, i.e., tone redundancy etc. The use of block codes that works better with burst errors alse has its advantages, i.e., Reed-Solomon.

If you wish, contact me by e-mail and I'll look for the titles of some technical papers and books that I have collected over the years that

I found useful. Its a very extensive topic.

Good luck!

--Johan, KC7WW

32. Re: [HFSIG:2865] Why so little participation in USA?

From: "Louis Cobet" <louis_cobet@email.msn.com>
Date: Sun, 25 Oct 1998 10:02:16 -0800
To: <hfsig@tapr.org>
Message-ID: <003001be0041$9bac7f00$c3242599@hp-customer>
Rendered body type: text/plain

I think there are many reasons but I want to talk about two.

One is the tech plus licensees (like me) can only send digital transmissions on ten meters. Not much traffic on ten for obvious reasons. That is 100,000 licensees. If only a few percent were interested that would make a difference. This category may have access after license restructuring.

Second, we need assembled bpsk units ready for use by any computer.

That is a challenge I would make. The software is pretty much written for most DSP chips. If someone would design a "black box" with standard RS232 in and Tx, Rx and PTT out, that would be a big start. A minimum cost design might be kitted by TAPR. Then maybe one of the ham manufacturers would license it.

Also, we need to talk up the advantages of bpsk in particular. Articles in all the ham magazines, presentations at the conventions, conferences and club meetings. National call frequencies are needed on all the bands. We should consider running contests. In short, we need to promote these new digital modes.

Louis Cobet, K6MDH
louis_cobet@email.msn.com

You wrote:
>Here is my pea:
>
>Are there still experimentors out there that are interested in the newer
>crop of digital modes for HF? Meaning actually getting on the air and
>playing with it -- helping with development work instead of just talking
>about it? What can be done to encourage participation?
>
>The bands are good right now -- equipment requirements are
>quite modest. For example; I run 40W output into a vertical antenna
>and using MTHELL or MT63 and the 56002EVM, can work
>VK2DSG with excellent results and regularity. These two modes are
>not critical as far as frequency accuracy is concerned. If you are
>within +/- 5 Hz, that is good. Most transceivers can do that.
>
>There also is PSK31, which now is avalailable for several platforms:
>56002EVM, TI320C50, SHARC EZ-KIT, and even Linux
>(using a sound card). This one works very well, even under poor
>conditions. It does, however require very very precise frequency
>tolerances, i.e., 2 Hz is about the limit. A rig like the ICOM 706 can
>do this quite easily, so that is no big drawback.
>
>As an alternative, VE2IQ's C-BPSK system also is a viable
>alternative. It has a somewhat-larger support group here in the USA,
>and regular trans-Atlantic contacts are commonly taking place. As
>for PSK31, frequency accuracy is most important. Equipment
>requirements otherwise being quite modest.
>
>But there also is the 15-tone AX.25 modem that will do TCP/IP over
>HF. It runs at a respectable 3800 bps (with error control 800 bps). I
>personally can attest to how well it works from channel simulator tests
>and over-the-air testing against worthy competitors like for example
>PACTOR II and CLOVER-2000. This modem, presently only runs
>on the 56002EVM. It is designed for application in a HF network
>topology and some kind of terminal program or NOS (KA9Q,
>JNOS, Linux, or equivalent) is needed to use or integrate it into
>existing networks. Knowledge of HF digital communications and
>computer networks are useful before trying this.
>
>These are just a couple of worthy pursuits --- Does anyone have
>any idea why there is so little participation here in the USA?
>

>Is it due to a lack of basic education, i.e.,
>"How-To" information ....
>possibly too little being published about it? ...
>Difficulty in getting the hardware together? ...
>Where to get the software? ...
>
>What can we do to attract more participation?
>
>73
>
>Johan, KC7WW
>
>P.S.
>For brevity, I did not even mention all the new digital stuff that
>is out there. I sincerely aplogise for that.
>

33. Re: [HFSIG:2872] Re: Coding i HF data transmission

From: malberts@ncs.com.na (Mike Alberts)
Date: Mon, 26 Oct 1998 08:59:32 +0200
To: hfsig@tapr.org
Message-ID: <36341DD4.21BA238D@ncs.com.na>
Rendered body type: text/plain

H all.

I am so much interested in Digital HF that I cannot tell how I would go about
to get going.....I need to get in touch with amateurs to give me
assistance...I live in Namibia V51 land and dont have all the facilities that
either the USA guys or the europeans amateurs have to their exposure....
73's
Mike
V51MA

Johan Forrer wrote:

> Halvor,
>
> Just a few thoughts ....
>
> A lot have been said about this topic in the past on this list. Of course
> as it relates to HF digital communications. It may be a good idea to
> search the archive (accessible from the TAPR web site) with a
> subject of "coding".
>
> One should think about a coding system that is taylored for a specific
> "channel" as something that works for HF probably will not
> necessarily work well on UHF. The coding has to do it work on
> different kinds of channel pertubations, some unique for the that
> channel.
>
> The main and major factor on HF channels is destructive multipath and
> fading affects. Second to that is noise behaviour, particularly a kind
> that may be modeled by Markovian statistics rather than the
> usual Gaussian model. The Watterson ionosperic propagation
> model is gereally accepted as a good one.
>
> With that bit of background, you want to design the coding system
> to effectively deal with signals that has randomly distorted bit edges,
> against a burst-noise background. What seems to work well are
> the use time- and frequency diversity techniques. Both bock and
> convolutional codes have their place, often in combination, i.e.,
> a convolutional inner code wrapped inside an appropiate block
> code. Time-diversity (interleaving) methods does wonderful things
> for HF, so does frequency diversity, i.e., tone redundancy etc.
> The use of block codes that works better with burst errors
> alse has its advantages, i.e., Reed-Solomon.
>
> If you wish, contact me by e-mail and I'll look for the titles of some
> technical papers and books that I have collected over the years that
> I found useful. Its a very extensive topic.
>
> Good luck!
>
> --Johan, KC7WW

34. Re: [HFSIG:2865] Why so little participation?

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Mon, 26 Oct 1998 19:49:01 +0000
To: hfsig@tapr.org
Message-ID: <3634D22D.68A1@marconicomms.com>
Rendered body type: text/plain

Some suggestions:

Set up some web pages defining the on-air standards for digital modes. The info I've seen is sketchy at best. Standards appear to be controlled by the whim of individuals, or the functions of some program someone once wrote. Set out a formal means by which the standards are updated. It won't stop anyone experimenting around with it, but for the sake of interoperability there has to be some controlled specifications.

Make library sourcecode available for the algorithms that implement the standard modes. That improves compliance and encourages porting.

Sell cheap `shink-wrapped' HF-modems, rather than `DSP EVM kits'. The idea of getting involved straight away with DSP probably puts some people off. A complete HF-modem can be built with 3 chips on a board area 2 square inches. I think we could match the high-street price tag of telephone line modems which ought to wake up a few people.

Has anyone considered if any of the current Flash-programmable telephone line modems could be re-programmed with HF modes? Many people have these already. Think 18 months' time, if previous form means anything, these will be old-hat and knocking around dirt cheap.

How about more emphasis on digital coded speech modes? Are people put off by the keyboard/text nature of the link? There's sourcecode for at least 2 speech coders usable on HF on the internet. Stick that on your DSP card...

The combination of amateur radio and interest in DSP as hobbies makes for a rather small group. Add to that the use of a specific DSP board, a specific computer and a specific OS. Anything to widen the resrictions would help.

eg. why write PC-specific host software? With free tools like Cygnus gcc and tcl/tk, it's easy to write programs that will compile and run on PC, Mac & Unix.

There doesn't seem to be an equivalent to receive only (DX) in digital modes. How about some digital broadcast stations? Perhaps it could transmit a digest of news and information culled from internet discussion relevant to HF digital operation?

How about setting up stations with a HF modem and a BBS? That would give people something to talk to and try out their gear 24 hrs a day, and create a rendezvous point for other contacts. As websites were what caused wired datacomms to take off, would it do the same for HF datacomms?

Many transceivers now use DSP in baseband processing, and sometimes support FSK. If there were put together a consistent package of digital waveform standards of good performance, and and a matching set DSP software implementations with wide support the transceiver makers may welcome the opportunity to ship with HF-modems built in. I have no doubt data facilities are in demand in semi-professional and commercial HF users.

Maybe it would enhance code portability and make the software accessible to more people if it were coded in C, and the sourcecode were more readily available. If I download something and unzip a .exe I ignore it - what's the point in trying something if I don't know what I tried? Far more people know C than DSP assembler code. If someone's got your sourcecode maybe they'll do you a favor and port it to their hardware.

The M56002 is obsolete, who would buy a 56002 EVM if they know anything about DSP? Do Motorola still sell EVMs? Float DSPs are much easier to program than fixed, and there's no cost penalty any more. Its no suprise that it's hard to persuade anyone to take up programming obsolete fixed-point DSPs as a hobby...

Maybe it's time for a DSP platform migration? I fancy the ARM-10
due out next year. 600 Mflops, 1 Watt @ 2V, gcc C compiler...

Anyway, so we get lots of black-box modem users on HF. Is that
what was intended? That equates to erp on the channels, yes.
Brainwaves occupied by HF modem technology, maybe not.

Just trying to be constructive &/or provocative

Ant
--
Suffering from `changed e-mail address syndrome'
____

A.N.Martin   E-mail anthony.n.martin@marconicomms.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
____

35. Re: [HFSIG:2875] Re: Why so little participation?

From: "Louis Cobet" <louis_cobet@email.msn.com>
Date: Mon, 26 Oct 1998 21:53:00 -0800
To: <hfsig@tapr.org>
Message-ID: <001e01be016e$0f194fe0$19befbd0@hp-customer>
Rendered body type: text/plain

All of the suggestions Anthony Martin made are good. All of the questions he
raised are good.

Do we want more people operating with these digital techniques or more
people participating in the design or coding of these digital techniques? I
think the two are interrelated.

I would like to see a hf modem developed, because I would be surprised if
more than at a maximum a few hundred people adopt the evaluation boards as
their working vehicle in ham radio. I do not think that will be a sufficient
critical mass to ensure an ongoing interest in the techniques in question.

One other thing, not mentioned so far is the need for beacons. Even
occasional one day beacons would help.

Louis Cobet, K6MDH
louis_cobet@email.msn.com

Anthony Martin wrote:
>Some suggestions:
> ....

36. Re: Why so little participation?

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Tue, 27 Oct 1998 07:20:51 +0100
To: hfsig@tapr.org
Message-ID: <36356643.A3A87B98@ife.ee.ethz.ch>
Rendered body type: text/plain

Anthony Martin wrote:

> Set up some web pages defining the on-air standards for digital
> modes. The info I've seen is sketchy at best. Standards appear to

I agree completely. While the source code is quite well documented,
there are easier ways to learn how these modes work :-)

And please don't implement the standard setting process the way
TAPR did with AX25 V2.2. Noone knew about the effort until
it was cast in stone, and the resulting standard is such
a horrible kludge that noone who implemented AX.25 protocol
stacks in the past is going to implement V2.2. So please
some public RFC process...

Tom

37. Re: [HFSIG:2876] Re: Why so little participation?

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Tue, 27 Oct 1998 04:16:48 -0600 (CST)

To: hfsig@tapr.org
Cc: 2258256@mcimail.com, Jack Mitchell <w5jam@hotmail.com>, wb5yqc@texas.net,
        rony@texas.net, Alan Cross <alan.cross@compaq.com>, w5edz@txdirect.net
Message-ID: <Pine.SOL.3.91.981027040708.12761B-100000@www.kelly-afb.org>
Rendered body type: text/plain

All,

Unless you get the ARRL behind any high speed modem effort, you won't find many hams involved in it and/or general acceptance of the effort by the ham community.

HQ ARRL staff hasn't seen fit to push it so you only choice is to use your influence with your ARRL Division Director. This might be a lost cause because many Division Directors are _not_ technical and/or are more interested in League politics.

I read the QST editorials by Dave Summer calling for more technology but it is an empty call...HQ won't support new technology...perhaps for fear that it will make some/one of the major equipment vendors mad and they won't send money advertising in QST.

That's my two cents worth.

Walt/K5YFW

```
    ========================================================================
    |           In a world without fences, who needs Gates?                |
    |----------------------------------------------------------------------|
    |        If I wanted "Windows", I'd knock a hole in my wall.           |
    ========================================================================
    |                                  |                                   |
    |                                  | The Micro$oft operating system    |
    | Walt DuBose - K5YFW              | didn't get as bad as it is        |
    | E-Mail k5yfw@www.kelly-afb.org   | overnight.  I has taken over 10   |
    | Business Telephone: (210)925-1171| years of careful, calculated,     |
    |     Home Telephone: (210)696-3196| development.                      |
    |                                  |                                   |
    ========================================================================
    |   Using MS is like eating high class dog food...its still dog food.  |
    ========================================================================
```

On Mon, 26 Oct 1998, Louis Cobet wrote:

> All of the suggestions Anthony Martin made are good. All of the questions he
> raised are good.
>
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> people participating in the design or coding of these digital techniques? I
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>
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> more than at a maximum a few hundred people adopt the evaluation boards as
> their working vehicle in ham radio. I do not think that will be a sufficient
> critical mass to ensure an ongoing interest in the techniques in question.
>
> One other thing, not mentioned so far is the need for beacons. Even
> occasional one day beacons would help.
>
> Louis Cobet, K6MDH
> louis_cobet@email.msn.com
>
> Anthony Martin wrote:
> >Some suggestions:
> > .....
>
>
>
>
>

38. Re: [HFSIG:2878] Re: Why so little participation?

From: "Andrew B. White" <k9cw@prairienet.org>
Date: Tue, 27 Oct 1998 07:57:17 -0600 (CST)
To: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Cc: hfsig@tapr.org
Message-ID: <Pine.3.89.9810270751.A27333-0100000@bluestem>
Rendered body type: text/plain

Walt,

Why do you think ARRL participation is a requirement? It seems
to me that the explosion of packet use started with the kits offered by
TAPR for the TNC1 and TNC2. Early AMTOR work started in the UK, not
here, although QST carried a couple construction articles. I don't
think that ARRL had anything to do with PACTOR-I or PACTOR-II or GTOR
development, and I know they had nothing to do with CLOVER-II development.

To become a popular ham digital mode, I think that there are a couple
requirements: it must use existing radios, it must be relatively
inexpensive, and there need to be other people on the air to work.

73, Drew

On Tue, 27 Oct 1998, Walt DuBose - K5YFW wrote:

> Unless you get the ARRL behind any high speed modem effort, you won't
> find many hams involved in it and/or general acceptance of the effort by
> the ham community.
>

------------------------------------------------------------------*
| Andrew B. White K9CW | internet: k9cw@prairienet.org |
| ABW Associates, Ltd. | phone/fax: 217-643-7327 |
------------------------------------------------------------------*

39. RE: Why so little participation in USA?

From: "Wheatley, Maurice" <mwheatle@trsystems.com>
Date: Tue, 27 Oct 1998 10:48:41 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <93C4EE270865D21185BF0000F80163EA01609F@WEBSERVER2>
Rendered body type: text/plain

My 2 cents,

What is the end goal of HF amateur radio experimentation? Is it to create
new improved modes for the entire amateur community or is it just for a
handful of experimenters to see what can be done with available technology?

Perhaps the common goal is what is missing. Currently new modes are
introduced by an individual(or company) with the hope that the ham community
will embrace their particular mode and create a de-facto standard.

If the end game is to create modes for widespread use, then I agree with Mr.
Martin that a common specification is agreed upon and common design goals
set before implementation and testing ever begins.

The first step would be to collect a "wish list" of what features are
desired by the amateur "market". This needs to take precedent over what is
technologically possible. A digital voice mode may be a cool technology but
had better have a distinct advantage over current analog voice modes to be
widely accepted by the masses.

The next step would be to sort out what can and can't be done and define
some general requirements for the new modes. The "fun" can then begin on
getting everyone to agree to a common spec. Once a spec is agreed upon then
it can be published for anyone to go develop using their favorite platform.

The biggest obstacle I think will be to get a consensus on a unified spec.
Hams seem to be very independent thinkers and somewhat inflexible in their
beliefs. I can envision many "religious' wars over wideband versus narrow,
coding A versus coding B, my great mode versus your stupid mode, etc.

I second the idea of beacon stations. Not only would they provide a good
source for the developers, but would also be a great way to promote the new
modes. A few articles in the major magazines, free software running on
common PC platforms to allow folks to listen in on the beacons with little
or no outlay of $$ could generate a lot of interest. Perhaps a "mode of the
week" where different experimental modes are tried and people can comment
back via email or reflectors like this one on how they work.

Hopefully there will be some renewed interest in this stuff. HFSIG has been
too quiet.

Tnx for the BW

Moe
AE4JY

40. Re: [HFSIG:2878] Re: Why so little participation?

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Tue, 27 Oct 1998 11:50:16 -0600 (CST)
To: "Andrew B. White" <k9cw@prairienet.org>
Cc: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.981027112630.13571A@www.kelly-afb.org>
Rendered body type: text/plain

Andrew,

You have identified the problem. In every case where a digital mode has "caught on", the ARRL has published articles supporting the mode. the more articles the publish in QST (the more they say) the more people turn to the mode. Its not where you get the hardware/software but obtaining support for the mode/project from the ARRL.

Back in the AM days (Before SSB) there was commercial SSB equipment and home brew stuff (DSB) but hams weren't buying it. It wasn't until the owner of Eldico got the SAC's commander General curt LeMay and his deputy Gen Griswall (?) to challange Art Collins and his gear at the Dayton Ham Fest back in the AM days and the ARRL's coverage of Gen LeMays Challange flight that SSB took off in the ham community. I remember working DSB (double slop bucket) and SSB (single slop bucket) with a home brew rig when AM operators would step all over you.

To get a small picture of the AM days, see the Nov 98 QST, page 47 "Zenith's 'One-and-Only' Ham Receiver".

Clover didn't get any recognition until QST ran an article on it...same with AX.25, AMTOR, Pactor, etc. How much interest is there in G-TOR (which I think is better than Pactor)...not much has been siad about it in QST to my knowledge...its just Kantronic's project as was Central Electronic's and Eldico's SSB project until QST followed the Lemay challange flight.

Its just marketing...there's no marketing for any of the high speed modems in QSt and TAPR's publication doesn't have the circulation (reach) that QST does.

The firm my wife used to work for took an unknown Mexican-American and got him elected mayor of San Antonio and then got him a position in the Clinton Cabinet...ever hear of Henry Cisernos?

Its all marketing....

Walt/K5YFW

```
    ========================================================================
    |           In a world without fences, who needs Gates?                |
    |----------------------------------------------------------------------|
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> *------------------------------*-------------------------------------*
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> | ABW Associates, Ltd. | phone/fax: 217-643-7327 |
> *------------------------------*-------------------------------------*
>
>

41. Re: [HFSIG:2880] RE: Why so little participation in USA?

From: Arvydas Vaisnys <arvydas.vaisnys@jpl.nasa.gov>
Date: Tue, 27 Oct 1998 10:07:41 -0800
To: hfsig@tapr.org
Message-ID: <3.0.32.19981027100740.008f6780@zorba.jpl.nasa.gov>
Rendered body type: text/plain

At 09:51 AM 10/27/98 -0600, you wrote:
>My 2 cents,
>
>What is the end goal of HF amateur radio experimentation? Is it to create
>new improved modes for the entire amateur community or is it just for a
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>back via email or reflectors like this one on how they work.
>
>Hopefully there will be some renewed interest in this stuff. HFSIG has been
>too quiet.
>
>Tnx for the BW
>
>Moe
>AE4JY

There is a big push to develop digital broadcasting in the bands below 30
MHz by the DRM: http://www.rnw.nl/DRM/index.html
If this effort succeeds there will be a lot of digital broadcasting in the
short wave bands. They are going through all the above stuff about
requirements, etc., and are working toward a "world standard" for digital

AM. Modem development so far is at the point where appriximately 20 kbps in
a 10 kHz channel appears feasible under some fairly severe channel
conditions. Both OFDM and single carrier systems are in contention. Modem
evaluations should take place early next year.

I have tried to think of ways the amateur radio community could be involved
in this development. Unfortunately the DRM is a closed organization, you
pay money to join. Seems like the radio amateur community's knowledge of
propagation would be a valuable asset to anyone developing and testing such
systems. If some beacons were set up, they could be used to provide
measurements of propagation parameters such as delay spread, which are
important for digital HF modem design.

Arv

42. RE: [HFSIG:2881] Re: Why so little participation?

From: "Lanier, Robert" <rlanier@harris.com>
Date: Tue, 27 Oct 1998 13:41:03 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <275399FB18C4D111871300805FBEB72F01215C18@corpmx6.ess.harris.com>
Rendered body type: text/plain

Well, I don't think it takes a rocket scientist to figure out why things catch
on when they appear in QST. The last time I checked, QST is read by more hams
and is published by the largest amateur radio organization in the world. With
that kind of exposure, its no wonder certain modes catch on the way they do.

I only wish they would publish more SS-related articles. Then maybe SS would
catch on more.

Tony KE4ATO

> -----Original Message-----
> From: hfsig@tapr.org [SMTP:hfsig@tapr.org]
> Sent: Tuesday, October 27, 1998 11:54 AM
> To: hfsig@tapr.org
> Subject: [HFSIG:2881] Re: Why so little participation?
>
> Andrew,
>
> You have identified the problem. In every case where a digital mode has
>
> "caught on", the ARRL has published articles supporting the mode. the
> more articles the publish in QST (the more they say) the more people
> turn
> to the mode. Its not where you get the hardware/software but obtaining
> support for the mode/project from the ARRL.
>
> Back in the AM days (Before SSB) there was commercial SSB equipment and
> home brew stuff (DSB) but hams weren't buying it. It wasn't until the
> owner of Eldico got the SAC's commander General curt LeMay and his
> deputy
> Gen Griswall (?) to challange Art Collins and his gear at the Dayton Ham
>
> Fest back in the AM days and the ARRL's coverage of Gen LeMays Challange
>
> flight that SSB took off in the ham community. I remember working DSB
> (double slop bucket) and SSB (single slop bucket) with a home brew rig
> when AM operators would step all over you.
>
> To get a small picture of the AM days, see the Nov 98 QST, page 47
> "Zenith's 'One-and-Only' Ham Receiver".
>
> Clover didn't get any recognition until QST ran an article on it...same
> with AX.25, AMTOR, Pactor, etc. How much interest is there in G-TOR
> (which I think is better than Pactor)...not much has been siad about it
> in QST to my knowledge...its just Kantronic's project as was Central
> Electronic's and Eldico's SSB project until QST followed the Lemay
> challange flight.
>
> Its just marketing...there's no marketing for any of the high speed
> modems in QSt and TAPR's publication doesn't have the circulation
> (reach)
> that QST does.
>
> The firm my wife used to work for took an unknown Mexican-American and
> got him elected mayor of San Antonio and then got him a position in the
> Clinton Cabinet...ever hear of Henry Cisernos?

Its all marketing....

Walt/K5YFW

```
=========================================================================
            |           In a world without fences, who needs Gates?
|

|-------------------------------------------------------------------------|
            |        If I wanted "Windows", I'd knock a hole in my wall.
|

=========================================================================
            |                                   |
|
            |                                   | The Micro$oft operating system
|
            | Walt DuBose - K5YFW               | didn't get as bad as it is
|
            | E-Mail k5yfw@www.kelly-afb.org    | overnight.  I has taken over
10 |
            | Business Telephone: (210)925-1171 | years of careful, calculated,
|
            |     Home Telephone: (210)696-3196 | development.
|
            |                                   |
|

=========================================================================
            |   Using MS is like eating high class dog food...its still dog
food. |

=========================================================================
```

> On Tue, 27 Oct 1998, Andrew B. White wrote:
>
> > Walt,
> >
> > Why do you think ARRL participation is a requirement? It seems
> > to me that the explosion of packet use started with the kits offered
> by
> > TAPR for the TNC1 and TNC2. Early AMTOR work started in the UK, not
> > here, although QST carried a couple construction articles. I don't
> > think that ARRL had anything to do with PACTOR-I or PACTOR-II or GTOR
> > development, and I know they had nothing to do with CLOVER-II
> development.
> >
> > To become a popular ham digital mode, I think that there are a couple
> > requirements: it must use existing radios, it must be relatively
> > inexpensive, and there need to be other people on the air to work.
> >
> > 73, Drew
> >
> >
> > On Tue, 27 Oct 1998, Walt DuBose - K5YFW wrote:
> >
> > > Unless you get the ARRL behind any high speed modem effort, you
> won't
> > > find many hams involved in it and/or general acceptance of the
> effort by
> > > the ham community.
> > >
> >
> > *-----------------------------*-------------------------------------*
> > | Andrew B. White K9CW | internet: k9cw@prairienet.org |
> > | ABW Associates, Ltd. | phone/fax: 217-643-7327 |
> > *-----------------------------*-------------------------------------*
> >
> >

43. News item

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Tue, 27 Oct 1998 18:48:04 +0000
To: hfsig@tapr.org
Message-ID: <36361564.1F@marconicomms.com>
Rendered body type: text/plain

Interesting excerpt from http://www-ixmp.itsi.disa.mil/1296min.htm

b. USN. Ms. Lisa Pinck, NRaD, briefed on the Automated Digital Network System (ADNS)
Battle Force E-Mail developed by the Navy. Ms. Pinck informed the Members that the Battle
Force E-Mail refers to the whole system and provides intership user to user service (E-Mail
messages, file attachments, etc.) and Automatic Transfer using standard HF or UHF radios.
ADNS consists of a hardware which includes a PC server and client Ethernet card; the NRaD-
designed serial E-Mail card; MIL-STD-188-110A compatible signal tone serial HF modem; and
connection into cryptos and radios. Its software consists of the Johanns Network Operating
System Amateur Radio server as a shareware, Eudora-Pro E-Mail as the client, and a Simple Mail
Transfer Protocol (SMTP)/POP3 compatible software. Ms. Pinck described the network and
server operation and mentioned how this system was developed primarily for small ships. Future
efforts include working on several battle groups where each group consists of 10 to 12 ships.

--
Suffering from `changed e-mail address syndrome'
_____

A.N.Martin E-mail anthony.n.martin@marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
_____

44. Re: [HFSIG:2879] Re: Why so little participation?

From: W6RCecilA <w6rca@ibm.net>
Date: Tue, 27 Oct 1998 14:22:54 -0700
To: hfsig@tapr.org
Message-ID: <363639AE.6B85@ibm.net>
Rendered body type: text/plain

Andrew B. White wrote:
> To become a popular ham digital mode, I think that there are a couple
> requirements: it must use existing radios, it must be relatively
> inexpensive, and there need to be other people on the air to work.

Allow me to add a third which would satisfy the first two: If it uses a standard Pentium PC and a standard sound card, almost all the requirements are met. A Pentium MMX has DSP capability. An SB-16 sound card is overkill for the A/D audio process. What is wrong with the PC platform?

This from a microcontroller/DSP ex-Intel employee. Anyone on internet has a computer. Many of those PCs are capable of digital data functions. Why not take advantage of it? Seems presently, all it would require is some DOS compatible applications interface software. What am I missing?
--
73, Cecil, W6RCA http://people.delphi.com/CecilMoore

45. FED-STD-1052 on-line

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Tue, 27 Oct 1998 21:48:38 +0000
To: hfsig@tapr.org
Message-ID: <36363FB6.1CBE@marconicomms.com>
Rendered body type: text/plain

See http://132.163.64.201/~bing/fs-1052/fs-1052.htm for an unusual treat. The FED-STD-1052 HF Modem standard in html.

FED-STD-1052 is a modem with the MIL-STD-188-110A waveform plus an ARQ protocol.

In the appendices it includes the 16-tone and 39-tone modem standards.

Enjoy!

Please come back to the "Re: Why so little participation?"
thread when you've finished; it's just getting interesting.
I don't want to divert anybody.

Ant
--
Suffering from `changed e-mail address syndrome'

A.N.Martin   E-mail anthony.n.martin@marconicomms.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK

46. Re: [HFSIG:2885] Re: Why so little participation?

From: Charles Brain <chbrain@dircon.co.uk>
Date: Tue, 27 Oct 1998 22:13:08 +0000
To: hfsig@tapr.org
Message-ID: <36364574.84BC8B74@dircon.co.uk>
Rendered body type: text/plain

W6RCecilA wrote:
>
> Allow me to add a third which would satisfy the first two: If it
> uses a standard Pentium PC and a standard sound card, almost all the
> requirements are met. A Pentium MMX has DSP capability. An SB-16
> sound card is overkill for the A/D audio process. What is wrong
> with the PC platform?
>
> is some DOS compatible applications interface software. What am I
> missing?

Hi,
 I don't think you are missing anything. It can be done even under
Windows.

 I have implemented a couple of DSP ish things using a P.C. The only
real problem I have found is some peculiarities of the Windows OS.

No matter how much buffering you put in there if you scroll your
windows about while your DSP application is running you will get
some latency problems. However for normal operation it is not a
problem.

Running in a DOS box gives you the same problem but it is not as bad.

Although I have done some soundcard programming on bare metal I now
prefer to use the Windows multimedia interface. (Its the only way I can
get my code to run on my Laptop).

Phil KA9Q has done some good work on using the Pentium FP processor in
assembler. Intel do the NSP DSP libraries (I used their FFT DLL in an
8ary FSK modem).

At the moment I am playing with the TI/Atlanta Signal Processing MELP
(Vocoder code). My little dialog application runs fine on a 166 Mhz
laptop in full duplex. As a comparison the TI Fixed point version
needs an 80 Mhz TMS320C50, however I am not sure what percentage of the
CPU it gobbles up.

- Charles

47. Re: Why so little participation?

From: k5cnf1@juno.com (Richard B Morrow)
Date: Tue, 27 Oct 1998 09:16:28 -0600
To: hfsig@tapr.org
Message-ID: <19981027.162328.6470.1.k5cnf1@juno.com>
Rendered body type: text/plain

Hey everyone,
having had my ticket for 43 years, I can tell you that the ARRL has
fought every new mode that came along from ssb to rtty and many other
things of this sort. Real old timers can remember the spark vs cw wars
that took place many years ago.

Even new antenna designs that did not "look" like they thought an antenna
should look. I knew some of the people that were fighting the ARRL over

rtty, as it was a more efficient traffic handling method than the cw nets were. Things like that.

I can remember that the number of solid state constructions articles that appeared in QST were very few until it became so apparent that tubes were sinking into the sunset. Even now, their diagrams are not quite standard. CQ and 73 have more technical articles than QST AND a most important point, the average ham now is NOT very technically smart anymore. Remember the tests were dumbed down, no more schematic drawing or written explanations of how things work.

Best bet is to hit all fronts. Then hope for the best.

CUL, K5CNF

_____
You don't need to buy Internet access to use free Internet e-mail.
Get completely free e-mail from Juno at http://www.juno.com/getjuno.html
or call Juno at (800) 654-JUNO [654-5866]

48. Fuzzy modes need operators!

From: "M. Greenman" <coombedn@ihug.co.nz>
Date: Wed, 28 Oct 1998 22:51:56 +1300
To: <hfsig@tapr.org>
Message-ID: <199810280952.WAA27925@smtp2.ihug.co.nz>
Rendered body type: text/plain

Hi folks,

There has been a debate recently about where all the digital mode activity has gone. Perhaps this will persuade some of you back on the air....

* What is Hellschreiber?
* Have you looked into operating Hellschreiber on HF at all?
* Do any of your present HF modes offer the ability to change speeds or fonts to show your individuality?
* Would you like to use foreign langage (non-English) fonts?
* Do you have trouble communicating on noisy bands, or with modest equipment?

Take a look at Hellschreiber!

There are two main forms in use - the traditional 1926 vintage Feld-Hell, which is on-off keyed like fast CW, and Multi-Tone Hell, which is a frequency domain mode. Both modes are UNCODED - which means that noise and other errors may distort the text, but cannot change it to anything else. These modes are designed to be read and interpreted by eye, not machine.

Both modes offer high performance, given the simplicity of equipment required. Both modes are very good at rejecting interference and are ideally
suited to noisy bands. MT-Hell also has a unique ability to reject QSB related timing errors and selective fading that plague FSK modes like RTTY -
MT-Hell requires no synchronism.

If you have an HF rig, and a Win95 computer with a sound card, you likely have all you need! There is also excellent software for the EVM56002 DSP kit.

There are about 30 - 40 Hell users world-wide at present, and world-wide contacts have been made, many under conditions where contacts in SSB and some other digital modes would not be possible. We desperately need more stations to talk to - so far most are in Europe, and there are only two in the US.

If you would like to learn more, take a look at the "Fuzzy Modes" web site:

http://members.xoom.com/ZL1BPU/Contents.html

Everything you need in the way of software is on the web site.

73,
Murray ZL1BPU

49. Information on packet high speed

From: "Nicola Savella" <savella@ieee.org>
Date: Wed, 28 Oct 1998 11:00:33 +0100

To: <hfsig@tapr.org>
Message-ID: <01be0259$cdba5c00$0100007f@vectra200-vl5>
Rendered body type: text/plain

Hallo everyone,
it is about 3 months that I'm reading emails in HFSIG mailing list, but I
have never writed to because I am not a amateur in a official way. In
december I will have the examination to become ham here in Italy.
I write to ask you some information, if someone can help me, he can wite me
also privately if the discussion is not interesting for all the mailing
list.
I'm am studing at university for my thesis the project of installing a PC in
a ambulance that can rx and tx files from a central station server. It is
required hig speed (Mbit/s) and strumentation not difficult to find in
PC-trade.
The main problem is now to search bibliografy on packet radio in HF and VHF,
can anyone tell me where to find these information?

Thanks everyone,
73 de Nicola
==
Nicola Savella
http://space.tin.it/io/nsavell
Biomedical Electronic Engineering - DEEI
University of Trieste - ITALY
Email: savella@ieee.org
==

50. Re: [HFSIG:2889] Fuzzy modes need operators!

From: malberts@ncs.com.na (Mike Alberts)
Date: Wed, 28 Oct 1998 12:20:46 +0200
To: hfsig@tapr.org
Message-ID: <3636EFFE.C530A9C3@ncs.com.na>
Rendered body type: text/plain

Hi Om's

Now a short note from Namibia.....
One can and will be able to get digital modes going if you access to
equipment...Unfortunately I live in Africa and the exchange rate KILLS all our
average Hams.....If you dont have thousands to blow...and dont have any
millionaire friend....SORRY you just stay behind....

I am lucky enoug to have 2 wonderfull american ham friens who helped me on the
way with equipment .....I am very keen to get digital modes going from Namibia
and have gone so far to get a PC going to set up a gateway...BUT without TNC's
etc and proper antenna's ...FORGET IT !!!!

Think what will I have to spent to import an antenna from the States ,,,Basic 3
element yagi....lets argue and say US$ 500.00 times the exchange rate which is
7.49 NS to 1 US $..Plus shipping and 125% import duties on electronic
equipment.....We "keep alive" on air with donated surplus equipment....No
average ham have 10.000 or more to get digital equipment in and up and running.

73
Mike V51MA

"M. Greenman" wrote:

> Hi folks,
>
> There has been a debate recently about where all the digital mode activity
> has gone. Perhaps this will persuade some of you back on the air....
>
> * What is Hellschreiber?
> * Have you looked into operating Hellschreiber on HF at all?
> * Do any of your present HF modes offer the ability to change speeds or
> fonts to show your individuality?
> * Would you like to use foreign langage (non-English) fonts?
> * Do you have trouble communicating on noisy bands, or with modest
> equipment?
>
> Take a look at Hellschreiber!
>
> There are two main forms in use - the traditional 1926 vintage Feld-Hell,
> which is on-off keyed like fast CW, and Multi-Tone Hell, which is a
> frequency domain mode. Both modes are UNCODED - which means that noise and
> other errors may distort the text, but cannot change it to anything else.
> These modes are designed to be read and interpreted by eye, not machine.
>

> Both modes offer high performance, given the simplicity of equipment
> required. Both modes are very good at rejecting interference and are
> ideally
> suited to noisy bands. MT-Hell also has a unique ability to reject QSB
> related timing errors and selective fading that plague FSK modes like RTTY
> -
> MT-Hell requires no synchronism.
>
> If you have an HF rig, and a Win95 computer with a sound card, you likely
> have all you need! There is also excellent software for the EVM56002 DSP
> kit.
>
> There are about 30 - 40 Hell users world-wide at present, and world-wide
> contacts have been made, many under conditions where contacts in SSB and
> some other digital modes would not be possible. We desperately need more
> stations to talk to - so far most are in Europe, and there are only two in
> the US.
>
> If you would like to learn more, take a look at the "Fuzzy Modes" web site:
>
> http://members.xoom.com/ZL1BPU/Contents.html
>
> Everything you need in the way of software is on the web site.
>
> 73,
> Murray ZL1BPU

51. Re: [HFSIG:2890] Information on packet high speed

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Wed, 28 Oct 1998 11:32:48 +0000
To: hfsig@tapr.org
Message-ID: <363700E0.3524@marconicomms.com>
Rendered body type: text/plain

Nicola Savella wrote:
>
> I'm am studing at university for my thesis the project of installing a PC in
> a ambulance that can rx and tx files from a central station server. It is
> required hig speed (Mbit/s) and strumentation not difficult to find in
> PC-trade.
> The main problem is now to search bibliografy on packet radio in HF and VHF,
> can anyone tell me where to find these information?

What you want to look at is the TETRA digital PMR system.
It will give the high data rates you want and is designed for the
needs of public services & utilities.

You won't get Megabits out of packet on VHF or HF. The idea of
packet is to make all stations store-and-forward repeater nodes,
whereas you're looking for a mobile-to-base architecture.

For your experiments a GSM mobile phone may do, or the Siemens GSM
modules. Athough TETRA offers higher data rates than GSM.

It seems unnecessary to re-invent the wheel when public services can
have this mobile data connectivity off-the-shelf. But they need people
like you to come up with the applications that use it. I guess your
system could access someone's medical records at the scene of an
accident
which seems like a great idea. It needs someone to demonstrate what
can be done - the people who run these organisations are often
ignorant of technology.

--
Suffering from `changed e-mail address syndrome'

A.N.Martin E-mail anthony.n.martin@marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK

---

52. Re: [HFSIG:2884] News item

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Wed, 28 Oct 1998 09:27:04 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.981028092155.15648F-100000@www.kelly-afb.org>

Rendered body type: text/plain

The Navy has been using HF E-mail for sometime. I believe that the
equipment is supplied by Harris RF Comm Group. The TCP/IP (SMTP) is
handled by a computer running SCO and they use a MIL STD 188-110(c) modem.

I understand that most data goes at 4800 BPS and in bad conditions at
2400 BPS.

This is far better than anything going on on the am bands and I'm 100%
sure that hams could duplicate or come very close to a MIL STD 188-110(c)
(39 tone parallel tone modem rather than serial tone modem) modem at a
resonable cost. Also, running TCP/IP (SMTP) on a 486 box running Linux is
very cost effective since the Linux OS is _FREE_.

Walt

```
    ========================================================================
    |           In a world without fences, who needs Gates?                |
    |----------------------------------------------------------------------|
    |        If I wanted "Windows", I'd knock a hole in my wall.           |
    ========================================================================
    |                                  |                                   |
    |                                  | The Micro$oft operating system    |
    | Walt DuBose - K5YFW              | didn't get as bad as it is        |
    | E-Mail k5yfw@www.kelly-afb.org   | overnight.  I has taken over 10   |
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    |     Home Telephone: (210)696-3196 | development.                     |
    |                                  |                                   |
    ========================================================================
    |   Using MS is like eating high class dog food...its still dog food.  |
    ========================================================================
```

On Tue, 27 Oct 1998, Anthony Martin wrote:

> Interesting excerpt from http://www-ixmp.itsi.disa.mil/1296min.htm
>
>   b.   USN. Ms. Lisa Pinck, NRaD, briefed on the Automated Digital
> Network System (ADNS)
> Battle Force E-Mail developed by the Navy. Ms. Pinck informed the
> Members that the Battle
> Force E-Mail refers to the whole system and provides intership user to
> user service (E-Mail
> messages, file attachments, etc.) and Automatic Transfer using standard
> HF or UHF radios.
> ADNS consists of a hardware which includes a PC server and client
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> connection into cryptos and radios. Its software consists of the
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> System Amateur Radio server as a shareware, Eudora-Pro E-Mail as the
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> the network and
> server operation and mentioned how this system was developed primarily
> for small ships. Future
> efforts include working on several battle groups where each group
> consists of 10 to 12 ships.
>
>
> --
> Suffering from `changed e-mail address syndrome'
> ____
>
> A.N.Martin   E-mail anthony.n.martin@marconicomms.com
> Snr. Development Engineer   Tel.   01245 353221 x4995
> Military Products Engineering   Fax.   01245 275993
> GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK
> ____
>
>

53. Re: [HFSIG:2885] Re: Why so little participation?

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Wed, 28 Oct 1998 09:39:03 -0600 (CST)
To: hfsig@tapr.org
Message-ID: <Pine.SOL.3.91.981028093608.15648L-100000@www.kelly-afb.org>

Rendered body type: text/plain

On Tue, 27 Oct 1998, W6RCecilA wrote:

> Andrew B. White wrote:
> > To become a popular ham digital mode, I think that there are a couple
> > requirements: it must use existing radios, it must be relatively
> > inexpensive, and there need to be other people on the air to work.
>
> Allow me to add a third which would satisfy the first two: If it
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> requirements are met. A Pentium MMX has DSP capability. An SB-16
> sound card is overkill for the A/D audio process. What is wrong
> with the PC platform?
>
> This from a microcontroller/DSP ex-Intel employee. Anyone on internet
> has a computer. Many of those PCs are capable of digital data functions.
> Why not take advantage of it? Seems presently, all it would require
> is some DOS compatible applications interface software. What am I
> missing?
> --
> 73, Cecil, W6RCA   http://people.delphi.com/CecilMoore
>
>

Let the software run on Linux as well as MS DOS.

Walt/K5YFW

```
=========================================================================
|           In a world without fences, who needs Gates?                 |
|-----------------------------------------------------------------------|
|        If I wanted "Windows", I'd knock a hole in my wall.            |
=========================================================================
|                                  |                                    |
|                                  | The Micro$oft operating system     |
| Walt DuBose - K5YFW              | didn't get as bad as it is         |
| E-Mail k5yfw@www.kelly-afb.org   | overnight.  I has taken over 10    |
| Business Telephone: (210)925-1171 | years of careful, calculated,     |
|     Home Telephone: (210)696-3196 | development.                       |
|                                  |                                    |
=========================================================================
|   Using MS is like eating high class dog food...its still dog food.   |
=========================================================================
```

54. Re: [HFSIG:2886] FED-STD-1052 on-line

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Wed, 28 Oct 1998 09:45:32 -0600 (CST)
To: hfsig@tapr.org
Cc: "Larry, W5UQ Higgins" <2258256@mcimail.com>,
Jack Mitchell <w5jam@hotmail.com>,
John Weber - WB5YQC <wb5yqc@texas.net>, w5edz@txdirect.net
Message-ID: <Pine.SOL.3.91.981028094027.15648N-100000@www.kelly-afb.org>
Rendered body type: text/plain

Sorry, I had forgotten about the FED-STD...this is what FEMA used I think...
better for hams to follow and already existing standard so we can
interface with FEMA/MARS, etc.

OBTW, if you haven't read the ham regs lately, Federal (and probably state/
local government) agencies can make first contact with hams directly on
ham frequencies in the case of an emeergency where loss of life and
properity is probable. Hummmmmm, and HF SMTP system where a Federal.
state, local government agency E-Mails hams notifying them of an emergency.
Of course the "agency" would probably be using the FED or MIL Std modem/
protocol.

Walt/K5YFW

55. Re: [HFSIG:2893] Re: News item

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Wed, 28 Oct 1998 13:49:22 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.4.05L.9810281345110.18824-100000@arctic>
Rendered body type: text/plain

On Wed, 28 Oct 1998, Walt DuBose - K5YFW wrote:

> The Navy has been using HF E-mail for sometime... equipment supplied

> by Harris RF Comm Group.... MIL STD 188-110(c) modem. (2400 baud)
>
> ... hams could duplicate or come very close to a MIL STD 188-110(c)
> (39 tone parallel tone modem rather than serial tone modem) modem at a
> resonable cost.

These modems originally cost over $50,000 but there are over 100 of them
floating around on the surplus market for under $125... Last I knew, Jim
Conrad <jjc@infi.net> had several in Norfolk, Va.

de WB4APR, bob

56. Re: [HFSIG:2893] Re: News item

From: "Harry L. Styron" <hlstyron@wcrklaw.com>
Date: Wed, 28 Oct 1998 10:59:28 -0800
To: hfsig@tapr.org
Message-ID: <3637698F.224C84DF@wcrklaw.com>
Rendered body type: text/plain

Walt DuBose - K5YFW wrote:

> The Navy has been using HF E-mail for sometime. I believe that the
> equipment is supplied by Harris RF Comm Group. The TCP/IP (SMTP) is
> handled by a computer running SCO and they use a MIL STD 188-110(c) modem.

So where does one get a copy of MIL STD 188-110(c)? And some reference to the
MODEM chip that was undoubtedly developed to implement it?

Harry Styron, K6MFV
[Attachment: vcard.vcf; text/x-vcard; 378 bytes]

57. RE: [HFSIG:2896] Re: News item

From: "Borcher, David (NG-EX)" <DBorcher@GI.com>
Date: Wed, 28 Oct 1998 14:13:20 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <0A7F08DCFAFFD1119FE6006097C510D05D6EA2@nogalesxch.nog.gi.com>
Rendered body type: text/plain

> ... hams could duplicate or come very close to a
MIL STD 188-110(c)
                        > (39 tone parallel tone modem rather than serial
tone modem) modem at a
                        > resonable cost.
        ****
        These modems originally cost over $50,000 but there are over 100 of
them
        floating around on the surplus market for under $125... Last I
knew, Jim
        Conrad <jjc@infi.net> had several in Norfolk, Va.

        de WB4APR, bob

Did you ever do anything with the ones you bought, Bob? Did the docs ever
show up?

Dave K7NG

58. Re: [HFSIG:2897] Re: News item

From: Charles Brain <chbrain@dircon.co.uk>
Date: Wed, 28 Oct 1998 19:21:10 +0000
To: hfsig@tapr.org
Message-ID: <36376EA6.74FEB600@dircon.co.uk>
Rendered body type: text/plain

Harry L. Styron wrote:

> So where does one get a copy of MIL STD 188-110(c)? And some reference to the
> MODEM chip that was undoubtedly developed to implement it?
>

That MIL standard contains 3 modems, a 16 tone one a 39 tone one
and a serial tone modem. The 39 Tone modem (110 Appendix B) that Bob is talking
about is duplicated in FED-STD 1052 Appendix A. Well in my copy it is.
I think someon said there was a copy of that FED-STD on-line.

I think you will find that it is more than a single modem chip. If the
units are more than a few years old they will be either bit slice processors
or implemented with analogue electronics.

- Charles

59. Re: [HFSIG:2898] Re: News item

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Wed, 28 Oct 1998 15:54:00 -0500 (EST)
To: hfsig@tapr.org
Cc: jjc@infi.net
Message-ID: <Pine.GSO.4.05L.9810281552140.18824-100000@arctic>
Rendered body type: text/plain

On Wed, 28 Oct 1998, Borcher, David (NG-EX) wrote:

> These modems originally cost over $50,000 but there are over 100 of them
> floating around on the surplus market for under $125... Last I knew,
> Jim Conrad <jjc@infi.net> had several in Norfolk, Va.
>
>   de WB4APR, bob
>   --
> Did you ever do anything with the ones you bought, Bob? Did the docs ever
> show up?

No, been to busy. I think that Jim made copies of the manual and that is
why the price went up. My $125 guess is only a GUESS!

60. Re: [HFSIG:2860] TAPR PSK Modem? (HTML Removed)

From: Lyle Johnson <lylej@azstarnet.com>
Date: Wed, 28 Oct 1998 18:55:25 -0700
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19981028185525.008e4340@pop.azstarnet.com>
Rendered body type: text/plain

See QEX for September 1987. May be available through the ARRL.

I have the docs here, but buried somewhere and not in electronic form.

Lyle

61. Re: Why so little participation?

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Thu, 29 Oct 1998 10:31:28 +0100
To: hfsig@tapr.org
Message-ID: <363835F0.F61C638B@ife.ee.ethz.ch>
Rendered body type: text/plain

Andrew B. White wrote:

> here, although QST carried a couple construction articles. I don't
> think that ARRL had anything to do with PACTOR-I or PACTOR-II or GTOR
> development, and I know they had nothing to do with CLOVER-II development.

> To become a popular ham digital mode, I think that there are a couple
> requirements: it must use existing radios, it must be relatively
> inexpensive, and there need to be other people on the air to work.

"inexpensive" is certainly not an attribute of Pactor II and Clover...

Tom

62. Re: [HFSIG:2899] Re: News item

From: "Harry L. Styron" <hlstyron@wcrklaw.com>
Date: Thu, 29 Oct 1998 06:04:04 -0800
To: hfsig@tapr.org
Message-ID: <363875D3.704BB220@wcrklaw.com>
Rendered body type: text/plain

Thanks for the comments:--

As an absolute newbie to this SIG, why does it not make better sense to use

use the DSP technology built into the Rockwell chips in the wire line modems?

Or simply start with the TS320 (I think) chips and implement 16 QAM or some-

thing along those lines?
 ----
o Harry L. Styron, Attorney at Law   o
o Tel: 925-932-4300 Fax: 925-944-0591 o
o "The devil is in the details."   o

---

63. Re: [HFSIG:2902] Re: Why so little participation?

From: W6RCecilA <w6rca@ibm.net>
Date: Thu, 29 Oct 1998 07:35:06 -0700
To: hfsig@tapr.org
Message-ID: <36387D1A.5908@ibm.net>
Rendered body type: text/plain

Thomas Sailer wrote:
> "inexpensive" is certainly not an attribute of Pactor II and Clover...

HAL ran a Christmas special on its P38 last year for $295.
--
73, Cecil, W6RCA http://people.delphi.com/CecilMoore

64. Re: Modem Standards

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Thu, 29 Oct 1998 16:00:42 +0000
To: hfsig@tapr.org
Message-ID: <3638912A.3F06@marconicomms.com>
Rendered body type: text/plain

In response to a number of issues raised:

> These modems originally cost over $50,000 but there are over 100 of them
> floating around on the surplus market for under $125... Last I knew,
> Jim Conrad <jjc@infi.net> had several in Norfolk, Va.
>

A new ARM-9401 or RF-5710 which do lots of modes including
MIL-STD-188-110A costs around 3000 UK pounds. Too much for hams
but not $50000 . If they did perhaps the US DoD could take more
care with the taxpayer's money!

The modems on the market today are designed with 5 year old DSP
technology. That means 2 DSPs, lots of RAM & Flash, dual-port
RAMs and a CISC processor. And a 20W SMPS. There's no `modem chip'.

To design now you would use a single DSP and the component cost
would be much less. Or a TMS320C54x or M563xx EVM card would do
it if you're into fixed-point. Otherwise, how about a pentium
motherboard?

> So where does one get a copy of MIL STD 188-110(c)? And some reference to the
> MODEM chip that was undoubtedly developed to implement it?

I posted the URL for FED-STAN-1052. Did you not see it?

http://132.163.64.201/~bing/fs-1052/fs-1052.htm

FS-1052 replicates the spec of the equalised coded
serial-tone modem in MIL-STD-188-110A and also the 16 & 39 tone
modems.

It only specifies how the transmitter works, which almost
any programmer could code from this spec. The receiver is left
as an exercise for the reader. The only thing that isn't
properly described is the channel filtering. IMO that's
an omission.

STANAG-4285 specifies a waveform very similar to MIL-STD-188-110A
but the spec is much more helpful and in the annexes it describes
various algorithms for the equaliser, their processing
requirements and performance comparisons. This information
can equally be applied to 110A. The channel filter is defined
in 4285 and is probably fine for 110A. There isn't any help on
sync detection, clock recovery, FEC decoding or how to achieve
the +-75 Hz doppler correction.

Prices for the documents in the UK quoted yesterday were
12.00 UKP for 4285 and 20.25 UKP for 110A. 4285 is a good
50 pages, it would cost most of that for copying and postage.
Both are unclassified.

But I've seen wider bandwidth modes attacked on this group
in favor of simple narrowband modes like BPSK. I'm suprised
at the interest.

If you think that 110A or 4285 uses too much BW (2800Hz) to
be usable under ham regs then STANAG-4529 is a
1240Hz BW modem of the same type. 4529 has shortcomings
and I'd suggest that a modem implementing the Canadian
comments against 4529 would be a better modem.

If it's low-signal poor-conditions you're working, the
110A 75bps mode has special treatment for just that. Although
unwritten in the spec, this mode was designed to have a much
longer DFE equaliser to work in exceptional conditions of high
multipath delay.

> Sorry, I had forgotten about the FED-STD...this is what FEMA
> used I think...
> better for hams to follow and already existing standard so
> we can interface with FEMA/MARS, etc.

I'm inclined to agree. Is there a need for yet more HF modem
standards with hams doing their own thing? If there are already
adequate standards, use em! If MIL_STD-188-110A is considered
to offer the height of performance then what's required is a
joint effort to develop public-domain C library code. People
can then turn that into implementations on their favorite
DSP/computer/OS platform without needing a PhD in mathematics.

You can still experiment with it, by adding longer equalisers
and such to enhance the receiver. You could chop out the fixed
patterns from the transmit waveform to deliver higher throughput
or more coding gain but only allowing DFE equalisers.
You can also follow-up with features like auto-bauding and
auto-interleaver detection and optional parts of the spec.

The only modem I know to have demonstrated higher performance
is the Enhanced Parallel Tone (EPT) developed by Stephen Cook
of the DSTO. This is an OFDM modem with rate 2/3 convolutional
coding. The on-air rate is 7200 bps giving a coded mode of
4800 bps. It's a bit fussy about the corners of the transceiver
channel filters as it uses every inch of BW. There's no
published standard for this waveform, although it is available
commercially in the ARM-9401. See Stephen Cook's paper:

http://www.itr.unisa.edu.au/~dstowww/longfish/hf95_6.html

(This link I got from hfsig a while back but it's so good
it's worth mentioning again for newbies.)

--
Suffering from `changed e-mail address syndrome'

A.N.Martin   E-mail anthony.n.martin@marconicomms.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK

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65. Re: Why so little participation?

From: Bob Johnson <bobj@cisi.com>
Date: Thu, 29 Oct 1998 10:24:42 -0500
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19981029102442.007748b8@mail.atlantic.net>
Rendered body type: text/plain

>To become a popular ham digital mode, I think that there are a couple
>requirements: it must use existing radios, it must be relatively
>inexpensive, and there need to be other people on the air to work.

And it must meet a need perceived by the user community.

If you increase throughput by 50%, will the average user care enough to go
out and spend $300 to get the new mode? Perhaps a new mode won't catch on
unless it is, say, three or five times as fast as what they are using now?

Or perhaps the average user would be more likely to jump on a new mode if
it allowed the same typical throughput as Pactor, but provided some minimum
throughput under conditions where Pactor fails completely? Or perhaps the
real need in the current user community is modes that make it faster and
easier to set up a link (hence the continuing popularity of Baudot RTTY)?

While there is no doubt that throughput sufficient to run HF TCP/IP would open lots of avenues for new development, the assumption that it is the MOST important factor in determining acceptance of a new mode has to be challenged.

And regardless of its technical merits, any new mode will not be accepted by the mainstream ham community until it works out-of-the-box. Having to build a system out of a DSP development board and a bunch of components is fun for me, but not for most of the hams I know, and these days I don't have _time_ for it either.

Finally, I've seen little evidence of ARRL staff taking sides in promoting new HF modes. All ham magazines are desperately short of good articles. You can tell this by the number of not-so-good articles they publish. If you have a favorite new mode, write a good article about it and send it to QST. APRS didn't catch on because of its technical brilliance: it started out as a rather limited hack (sorry Bob -- keep reading) that succeeded because

(1) Bob Bruninga didn't spend years sitting around and theorizing about how to design the absolute best system: he developed something that met the immediate needs of a lot of hams and started using it.

(2) He didn't wait for the ARRL to "discover" his system and declare it a "standard". He wrote articles explaining the advantages of his system and sent them to QST (and others).

And, of course, Bob and the other APRS developers are now refining the system into something closer to the "perfect" system that many of us would have aimed for in the first place. The point is that it doesn't matter how technically brilliant a new mode is: it only matters that the end user perceive it as useful to them (the secret of Microsoft's success). With that in mind, I offer some additional purely speculative qualities that might make a new mode desirable in the user community:

1) Look at how Pactor is used -- much of what I run across is individuals downloading mail from HF mail boxes. They spend as much time waiting for and reading the responses from the host, as they do waiting during the actual data transfer. Perhaps a mode that allows multiple users to share a frequency is actually what these people need?

2) Obviously, at HF you can't put too many people on one frequency and still get decent throughput. Perhaps a system that automatically moves a user to an "empty" frequency once a link has been established is the way to go (it's already in use in the marine industry). The problem is, of course, that supporting a mailbox host system that can simultaneously operate on multiple frequencies becomes very expensive.

3) Aha! In that case, maybe what we need is a mode that works nearly as well as Pactor, but for which a dedicated transceiver can be purchased for, say, $250, making it affordable for one computer to be talking to several radios simultaneously. Or perhaps we need a mode that lends itself to multiplexed operation through a single transceiver. The current popular crop of modes were designed to work with SSB transceivers because that's what people owned, but maybe it's time to reexamine that requirement.

-- Bob Johnson
WB4JCM

66. Re: [HFSIG:2906] Re: Why so little participation?

From: Charles Brain <chbrain@dircon.co.uk>
Date: Thu, 29 Oct 1998 17:47:18 +0000
To: hfsig@tapr.org
Message-ID: <3638AA26.BA069A62@dircon.co.uk>
Rendered body type: text/plain

>
> And it must meet a need perceived by the user community.
>

Unless the user community does not know it needs it. Lets face it entrepreneurs make their money by producing things people did not even know they wanted!

> If you increase throughput by 50%, will the average user care enough to go

Yes I agree, probably not.

> While there is no doubt that throughput sufficient to run HF TCP/IP would
> open lots of avenues for new development, the assumption that it is the

This technology already exists (newqpsk) maybe it hasn't taken off because
you need an EVM to run it. See Johans short article in PSR.

> Finally, I've seen little evidence of ARRL staff taking sides in promoting
> new HF modes. All ham magazines are desperately short of good articles.

If you look in Sept QST page 54 section 41 you will see that the ARRL future
systems committee are investigating the bandwidth/speed issue on HF amongst
other things. I seldom read Moved and Seconded section! So maybe more attention
should be paid to ARRL politics.

If wasn't for the legality issue (i.e broadcasting is illegal on the amateur bands
in the U.K other than for CQ calls and specially licensed stations). I would say a
broadcast mode with heavy FEC based on P.C technology is needed. Something like
Amtor Mode B but with deeper interleaving and much more powerful FEC. This could
be used for W1AW broadcasts, RSGB news broadcasts DGPS. And is something that SWLs
could participate in using. Things like Clover pretty much exclude listeners.

- Just some thoughts
- Charles G4GUO

67. Re: [HFSIG:2906] Re: Why so little participation?

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Thu, 29 Oct 1998 21:54:26 +0000
To: hfsig@tapr.org
Message-ID: <3638E412.36D@marconicomms.com>
Rendered body type: text/plain

Bob Johnson wrote:
>
> >To become a popular ham digital mode, I think that there are a couple
> >requirements: it must use existing radios, it must be relatively
> >inexpensive, and there need to be other people on the air to work.
>
> And it must meet a need perceived by the user community.
>
> If you increase throughput by 50%, will the average user care enough to go
> out and spend $300 to get the new mode? Perhaps a new mode won't catch on
> unless it is, say, three or five times as fast as what they are using now?
>

I thought the problem was that the average user doesn't
have a modem for any mode. And they're not interested
in digital HF. Either through ignorance, a perception
that it's too difficult or without merit or that there's
no-one to talk to, or through cost. Which I don't know.

But if they know what amazing things HF datacomms
can do and they decide to stick with morse because
they like it, that's fine.

> Or perhaps the average user would be more likely to jump on a new mode if
> it allowed the same typical throughput as Pactor, but provided some minimum
> throughput under conditions where Pactor fails completely? Or perhaps the
> real need in the current user community is modes that make it faster and
> easier to set up a link (hence the continuing popularity of Baudot RTTY)?

I don't think changing users from one mode to
another will increase participation. Unless you
give them slower modems so they spend longer
at it :-) You need non-users to become users.

>
> While there is no doubt that throughput sufficient to run HF TCP/IP would
> open lots of avenues for new development, the assumption that it is the
> MOST important factor in determining acceptance of a new mode has to be
> challenged.

A modem designed for internet would have a slow connect, fast
over-over without resync and a cleardown. HF modems aren't
designed that way but should be. It should avoid interleaver
delays in favour of ARQ. I think it's better to write your
own sockets driver and hang your internet apps on that,
rather than try to handle packets after TCP/IP has had a go.
Take the HTTP protocol; that's ideal for HF, it's lock-step
simplex and clients will wait ages without being offended.

FTP, SMTP and the rest are the same. Something is still
needed for the virtual ports, addressing, and ARQ, just
not TCP.

Hey, Is anyone at this moment getting hfsig over HF?

If so I aplogise for the length of this post...

<snip stuff about Bob>

I'm all in favor of go-it-alone.

And people have to have a use for this digital stuff.
If its just another way of taking one-to-one then no-ones
going to get it. If you can get that CCD camera and send
them a jpeg file of you, that's different. Or collect your
mail, or download some ham software, read the noticeboard
and reports of who's been logged in from where, and again
that's different. Something you can't do with speech or
morse.

>
> 1) Look at how Pactor is used -- much of what I run across is individuals
> downloading mail from HF mail boxes. They spend as much time waiting for
> and reading the responses from the host, as they do waiting during the
> actual data transfer. Perhaps a mode that allows multiple users to share a
> frequency is actually what these people need?

I thought there wasn't much participation? You're saying that you run
across them all the time. And you're saying the channel's too busy!

You have FDMA, TDMA and CDMA or CSMA/CD. Take your pick.

The trick is to get the data across as fast as possible
on a 1st come 1st served basis and free the channel for
others. That minimises the total waiting of the users. The
less time you're on, the less chance of collisions. Speed
is an important issue here.

>
> 2) Obviously, at HF you can't put too many people on one frequency and
> still get decent throughput. Perhaps a system that automatically moves a
> user to an "empty" frequency once a link has been established is the way to
> go (it's already in use in the marine industry). The problem is, of
> course, that supporting a mailbox host system that can simultaneously
> operate on multiple frequencies becomes very expensive.

Quite obviously if a server must support 10 clients it needs 10 times
the data rate available to it than each client. That does rather make
the economics difficult for the bloke who owns the server.

Just rely on people not using it too often. If you get a lot of
customers consider raising a small amount of money from each of
them to pay for another server system. That way it grows. Otherwise
it gets chock-a-block and people turn off their modems.

> 3) Aha! In that case, maybe what we need is a mode that works nearly as
> well as Pactor, but for which a dedicated transceiver can be purchased for,
> say, $250, making it affordable for one computer to be talking to several
> radios simultaneously. Or perhaps we need a mode that lends itself to
> multiplexed operation through a single transceiver. The current popular
> crop of modes were designed to work with SSB transceivers because that's
> what people owned, but maybe it's time to reexamine that requirement.
>

A multi-channel transmitter and receiver is a feasible idea. Digitize
several channels together and put a big fat DSP on the back running
several modems. The server must exercise some form of power control
over the clients to get similar received levels in all channels
otherwise
linearity / dynamic range of the ADC is a problem. A nice exercise there
for you in software radio design. Is it legal? Let us know how you get
on!

> -- Bob Johnson
>   WB4JCM

Yes I get your main point: any standard is better than none.
It doesn't have to be the ultimate in performance, just be
easily accessible and promoted in the right way.

We also need to DO something different, however small, in
order to write about it & enthuse others. And we need good
writers & publicists, not just DSP gurus.

Ant
--
Suffering from `changed e-mail address syndrome'

A.N.Martin   E-mail anthony.n.martin@marconicomms.com
Snr. Development Engineer   Tel.   01245 353221 x4995
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GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK

68. Re: [HFSIG:2908] Re: Why so little participation?

From: Bob Johnson <bobj@cisi.com>
Date: Thu, 29 Oct 1998 20:37:03 -0500
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19981029203703.0076bc9c@mail.atlantic.net>
Rendered body type: text/plain

At 03:56 PM 10/29/98 -0600, Anthony Martin wrote:
>Yes I get your main point: any standard is better than none.
>It doesn't have to be the ultimate in performance, just be
>easily accessible and promoted in the right way.

Well, that's part of my point. The other part (other point?) is that all
of the proposals for new modes that I see are concerned solely with
increasing throughput in point-to-point links. That's a wonderful goal,
and is necessary and sufficient for some applications, but we need to step
back once in a while and ask whether some other factors explain the
relative lack of interest in HF digital modes, particularly new modes. As
you mentioned, developing transport (and routing) protocols tailored for
the environment is probably an important part of the solution.

>We also need to DO something different, however small, in
>order to write about it & enthuse others. And we need good
>writers & publicists, not just DSP gurus.
>

Exactly! The best way to get more poeple interested in using new modes is
to let more people know they exist.

-- Bob Johnson
   WB4JCM

69. Re: Why so little participation?

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Fri, 30 Oct 1998 12:05:34 +0000
To: hfsig@tapr.org
Message-ID: <3639AB8E.4B95@marconicomms.com>
Rendered body type: text/plain

Thomas Sailer wrote:
>
> > delays in favour of ARQ. I think it's better to write your
> > own sockets driver and hang your internet apps on that,
> > rather than try to handle packets after TCP/IP has had a go.
>
> One usually calls that a protocol family :-)
>
> The downside of this would be that you need to adapt
> applications, as they are usually quite hardcoded
> to the INET protocol family. And also, you can't do any
> routing.
>
> On the other hand, schemes like Van Jacobson header
> compression and selective dropping of packets are pretty
> efficient in taming a TCP/IP stack.
>
> Tom

I can't see how you would be able to get hold of the
TCP/IP packets either without having to alter the

operating system. Once they're formatted, if they're not
addressed to yourself they inevitably go straight out
of the machine on ethernet/slip/ppp!

How would you propose hijacking the TCP/IP packets?

At the socket level, internet access is very simple.

Anyone can write a simple client or server.

Client applications normally access TCP/IP sockets
via 5 main C calls:
connect()   bind() read() write() close()
On unix they may also use:
select() or poll()
A server may additionally use:
accept()

Applications also use a call to access the name lookup
to translate hostnames into IP addresses. On Windows
the functions may be called something else. This is
is because socket streams, file streams, com port sreams,
and audio streams all look quite different to each
other on windows. On Unix they all look the same.

The socket API is exactly the same for INET or X25
or any other protocol stacks. What's to stop us
stacking AX25 behind it, and pretending it's INET?
Netscape will know nothing about what we did.

Sockets are a facility added to PCs via DLLs. Just write
a new DLL, load that instead and netscape will be happily
talk to it. (not being a PC guru, I can't be sure about
that.)

On Unix you could do it by fiddling with libc.
The alternative is recompiling the apps with
the addition of an extra `spoof' socket
library. and for most Unix apps that's not hard,
it's just plain boring, business as usual.

The implementation of `SOCKS' is an example of code
that `steals' the sockets calls of applications
running on machines with no internet access and
remotes them onto machines which do have internet
access and a `SOCKS' server (sockd). Netscape has
`socks' already compiled in, and most apps are
readily `socksified'.

One wangle that comes to mind is to butcher `sockd',
splitting it in two parts with the HF modem link in
the middle. That will work for any existing application
that supports SOCKS. You type a URL in netscape,
netscape hands it to sockd on your own machine, it
goes via HF to sockd on a remote server, and there
it gets dropped into TCP/IP. A HTTP server can then
pick up the request and the reply goes back in the
reverse direction.

Sounds so good I might just give it a go.

ps. where can I get source for an AX-25 stack?

--
Suffering from `changed e-mail address syndrome'

A.N.Martin   E-mail anthony.n.martin@marconicomms.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK

---

70. Re: Why so little

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 30 Oct 1998 13:57:28 +0000
To: hfsig@tapr.org

Message-ID: <3639C5C8.9D38A640@dircon.co.uk>
Rendered body type: text/plain

Anthony Martin wrote:
>
> Sounds so good I might just give it a go.
>

The sad thing about practically everything that has been
discussed is that it has already been done, it is just that
few people are aware of the work.

This Internet idea, for instance, in the Linux environment
AX.25 can now be compiled into the kernel. For Windows a
guy in Greece has written a pseudo modem i.e it appears as
a dialup networking object on your P.C and implements AX.25
This software works with standard HTML browsers.

The relatively high speed HF modems exist as I mentioned earlier.

It all seems to boil down to what was said before, the packages need
to be shrink wrapped and publicized.

Or maybe it is just that as far as data enthusiasts are concerned the
Internet is more reliable and fun than H.F and goes further than VHF.

- Charles

71. Re: Why so little participation?

From: "Fernando M. I. Carreiro" <fmi@carreiro.com>
Date: Fri, 30 Oct 1998 19:17:08 -0000
To: <hfsig@tapr.org>
Message-ID: <001a01be043a$1450b380$e6ffa8c0@pc001>
Rendered body type: text/plain

Greetings to all,

    I've been following this thread for a while now, and I've been trying hard not to join the
discussion. However after the following comment, I decided that it was time that I did:

    > From Charles Brain:
    > It all seems to boil down to what was said before, the packages
    > need to be shrink wrapped and publicised.

    Well, believe or not, that is what I am currently doing. However, the product, is still not ready to
 be launced or publicised. I am only mentioning it now just to settle some discussion issues.

    The project I am working on, goes from the Modem Unit right up to computer Applications for various
functions such as Internet Browsers, Digital Voice, E-Mail, and others. It will also cater for various
protocols such as AX.25, Compressed AX.25, TCP/IP, DLC, NetBEUI, IRDA (yes that's the infrared
protocol). The modem is capable of 15-17kbps over HF/VHF/UHF SSB/AM/FM (yes kilo bits per sec). The
modulation is such that it is capable of the high speeds both on HF as VHF (saving money one modem fits
all environments).This rate was chosen so to provide hardware modems as well as software modems using
sound cards on a PC. With higher DSP clock speeds, the hardware unit can reach 50kbps. Here are some
details:

    Modem Types: Closed-box Hardware or Software Modem for Sound Card (to cater for both markets)
    Modulation: 6 tone, 4VPSK or 5VPSK modulation with Doppler Correction.
    Sampling: 44.1kHz
    RAWt Rate: 15151kbps at 4VPSK, or 17222kbps at 5VPSK
    Cooked Rate: 11363kbps, or 12917kbps (using 25% ECC redundancy)
    Format: MPT1327 with error correction (64bit codeword = 48 bit data, 16 bit ECC)
    Interface: RS-232C KISS protocol (PPP at a later stage)
    Price Range:
        Hardware modem: $300 - $400 for prototype. Production model will be lower.
        Software modem: $70 - $80 for prototype. Production model will be lower.

    Protocol Stacks:
        Windows '95/'98/NT:
            Interface and Network Protocol stacks drivers, at the Operation System level.
            Functions just like a standard Network Adapter Driver and Protocol Stack.
            AX.25 (Standard v2.2 AX.25 protocol)
            CAX.25 (Compressed version of AX.25)
            RadioNET (Special High speed protocol, specific for this modem, including routing)
            other protocols are provided standard by OS, such as TCP/IP, NetBEUI, IPX/SPX, DLC, IRDA
            including routing for IPX/SPX and TCP/IP
        Linux
            Drivers at the OS level as the other already standard protocols.
            CAX.25 (Compressed version of AX.25)

RadioNET (Special High speed protocol, specific for this modem, including routing)
other protocols are provided standard by OS, such as TCP/IP, AX.25
Others
For development after launch. Will be relying on contribution by other HAMs.
Mac's, UNIX, VMS, etc. DOS was not contemplated due to the development requirements
for all the aspects of software for protocols, applications, etc. However, I will provide details
and source code licenses for all that wish to port the software for all other OS environments.

Applications
Since the protocol stacks are at the OS level, all software for that OS will work.
Internet functions,Network functionality, Chat, File sharing, Database services,
However specific applications will provide special functions for HAM environment
such as Digital voice communications, Broadcast Chat, Beacons, logging,
and identification of stations in range, etc.

Publicity
When completed and launch, details will be posted to Mail-lists, a Web Site setup,
Articles will be submitted to TAPR, and ARRL, etc.
Complete technical details of modem and software will be placed on the internet.
Source will also be available to licensed users depending on there requirements.
Direct Sales will be handled on the Web Site.

Well, this is just to let you know there are some HAM's participating, but maybe not at a large scale or in large numbers. All development has been done in the quite until now, because of personal reasons. However if a reasonable number of HAM's really want to get access to more info, I will open a discussion thread. But please note, that I will not able to give too much attention to the discussions, as I have to give it to the project in order to complete it as soon as I can. Although I have a couple of HAM's aiding in tests and experiments, all development is a one-man job. Those that wish to participate in Beta testing should submit the details to me directly, although I am still in the early stages of the alpha testing. It should still be a while before going into beta testing. Production and Commercialisation will be handled by a company in Portugal, since this is where I am living at the moment, even though I have a South African callsign. I will be changing to a Portuguese callsign soon.

Too all, thank you and kind regards,

Fernando M. I. Carreiro
mailto:fmi@carreiro.com
callsign:ZS6BLF

72. Re: [HFSIG:2910] Re: Why so little participation?

From: Eduardo Jacob <edu@kender.es>
Date: Fri, 30 Oct 1998 20:51:35 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19981030205135.00869210@kender.es>
Rendered body type: text/plain

>ps. where can I get source for an AX-25 stack?

From Linux AX-25 utils and Linux Kernel. New 2.1 is even more ham friendly. More: I guess that Linux is the Ham friendlinest operating system ever written. And everything with source.

Eduardo

73. Re: [HFSIG:2912] Re: Why so little participation?

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 30 Oct 1998 20:23:30 +0000
To: hfsig@tapr.org
Message-ID: <363A2042.7759981@dircon.co.uk>
Rendered body type: text/plain

Fernando M. I. Carreiro wrote:
>
> Well, believe or not, that is what I am currently doing.

Well I wish you luck with your product.

> The modem is capable of 15-17kbps over HF/VHF/UHF SSB/AM/FM
> Format: MPT1327 with error correction (64bit codeword = 48 bit data, 16 bit ECC)

I will be pleasantly suprised if you can achieve that data rate
on HF with current amateur equipment with its poor group delay
performance and HF with its severe multipath and interference levels.

- Charles

74. Re: [HFSIG:2911] Re: Why so little

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Fri, 30 Oct 1998 21:41:37 +0000
To: hfsig@tapr.org
Message-ID: <363A3291.2FAA@marconicomms.com>
Rendered body type: text/plain

Charles Brain wrote:
>
> The sad thing about practically everything that has been
> discussed is that it has already been done, it is just that
> few people are aware of the work.
>

Thanks Charles.

I don't mind looking stupid. It demonstrated a point.

OK so technology and standards aren't the problem. It's the
fact that no-one tells anyone about anything.

Oh it really is so so sad that no-one knows about this.

Maybe I'm too narrow-minded and I only read the HF modem stuff.
Perhaps I ought to read the VHF and microwave stuff too.
Nobody ever talks about AX-25 on hfsig.

Eduardo Jacob wrote:
>
> >ps. where can I get source for an AX-25 stack?
>
>   From Linux AX-25 utils and Linux Kernel. New 2.1 is
> even more ham friendly. More: I guess that Linux is the
> Ham friendlinest operating system ever written. And everything
> with source.
>
There's no excuse really for hams not using linux. If they want it
shrinkwrapped, you couldn't get better than that. Install the OS
and it's all-inclusive and ready-to-go.

OK Eduardo, so where do I get it? I'm using Sun Unix, so I don't
frequent the linux places.

I've got a freebie SuSE Linux 5.2 CD in my briefcase.
Is it on there somewhere? Oh no, I don't believe this. I've
been carrying it around and yet I didn't know I had it.

So enlighten me chaps, which of the HF modem implementations
mentioned on hfsig will actually work with this sweet linux
AX-25 stack :-)

--
Suffering from `changed e-mail address syndrome'

A.N.Martin   E-mail anthony.n.martin@marconicomms.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
GEC-Marconi Communications Ltd. Marconi House, New St. Chelmsford, UK

75. Re: [HFSIG:2912] Re: Why so little participation?

From: Walt DuBose - K5YFW <k5yfw@www.kelly-afb.org>
Date: Fri, 30 Oct 1998 16:08:38 -0600 (CST)
To: hfsig@tapr.org
Cc: "Larry, W5UQ Higgins" <2258256@mcimail.com>,
        Jack Mitchell <w5jam@hotmail.com>,
        Jack Mitchell <w5jam@total-access.net>,
        John Weber - WB5YQC <wb5yqc@texas.net>, w5edz@txdirect.net,
        rony@texas.net
Message-ID: <Pine.SOL.3.91.981030160450.21025A-100000@www.kelly-afb.org>
Rendered body type: text/plain

Fernandom,

Very interesting...looks like just what the doctor ordered.

Just a couple of quick questions...1) what is the maximum baud rate (or

phase shift) of the transmitted signal and 2) what is the signal bandwidth?

Walt/K5YFW

Walt DuBose - K5YFW	The Micro$oft operating system didn't get as bad as it is overnight. I has taken over 10 years of careful, calculated, development.
E-Mail k5yfw@www.kelly-afb.org	
Business Telephone: (210)925-1171	
Home Telephone: (210)696-3196	

On Fri, 30 Oct 1998, Fernando M. I. Carreiro wrote:

> Greetings to all,
>
> I've been following this thread for a while now, and I've been trying hard not to join the discussion. However after the following comment, I decided that it was time that I did:
>
> > From Charles Brain:
> > It all seems to boil down to what was said before, the packages
> > need to be shrink wrapped and publicised.
>
> Well, believe or not, that is what I am currently doing. However, the product, is still not ready to be launced or publicised. I am only mentioning it now just to settle some discussion issues.
>
> The project I am working on, goes from the Modem Unit right up to computer Applications for various functions such as Internet Browsers, Digital Voice, E-Mail, and others. It will also cater for various protocols such as AX.25, Compressed AX.25, TCP/IP, DLC, NetBEUI, IRDA (yes that's the infrared protocol). The modem is capable of 15-17kbps over HF/VHF/UHF SSB/AM/FM (yes kilo bits per sec). The modulation is such that it is capable of the high speeds both on HF as VHF (saving money one modem fits all environments).This rate was chosen so to provide hardware modems as well as software modems using sound cards on a PC. With higher DSP clock speeds, the hardware unit can reach 50kbps. Here are some details:
>
> Modem Types: Closed-box Hardware or Software Modem for Sound Card (to cater for both markets)
> Modulation: 6 tone, 4VPSK or 5VPSK modulation with Doppler Correction.
> Sampling: 44.1kHz
> RAWt Rate: 15151kbps at 4VPSK, or 17222kbps at 5VPSK
> Cooked Rate: 11363kbps, or 12917kbps (using 25% ECC redundancy)
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> Interface: RS-232C KISS protocol (PPP at a later stage)
> Price Range:
> Hardware modem: $300 - $400 for prototype. Production model will be lower.
> Software modem: $70 - $80 for prototype. Production model will be lower.
>
> Protocol Stacks:
> Windows '95/'98/NT:
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> Functions just like a standard Network Adapter Driver and Protocol Stack.
> AX.25 (Standard v2.2 AX.25 protocol)
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> other protocols are provided standard by OS, such as TCP/IP, NetBEUI, IPX/SPX, DLC, IRDA
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> Linux
> Drivers at the OS level as the other already standard protocols.
> CAX.25 (Compressed version of AX.25)
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> other protocols are provided standard by OS, such as TCP/IP, AX.25
> Others
> For development after launch. Will be relying on contribution by other HAMs.
> Mac's, UNIX, VMS, etc. DOS was not contemplated due to the development requirements
> for all the aspects of software for protocols, applications, etc. However, I will provide details
> and source code licenses for all that wish to port the software for all other OS environments.
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> Applications
> Since the protocol stacks are at the OS level, all software for that OS will work.
> Internet functions,Network functionality, Chat, File sharing, Database services,
> However specific applications will provide special functions for HAM environment
> such as Digital voice communications, Broadcast Chat, Beacons, logging,
> and identification of stations in range, etc.
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> Publicity
> When completed and launch, details will be posted to Mail-lists, a Web Site setup,

> Articles will be submitted to TAPR, and ARRL, etc.
> Complete technical details of modem and software will be placed on the internet.
> Source will also be available to licensed users depending on there requirements.
> Direct Sales will be handled on the Web Site.
>
>
> Well, this is just to let you know there are some HAM's participating, but maybe not at a large scale or in large numbers. All development has been done in the quite until now, because of personal reasons. However if a reasonable number of HAM's really want to get access to more info, I will open a discussion thread. But please note, that I will not able to give too much attention to the discussions, as I have to give it to the project in order to complete it as soon as I can. Although I have a couple of HAM's aiding in tests and experiments, all development is a one-man job. Those that wish to participate in Beta testing should submit the details to me directly, although I am still in the early stages of the alpha testing. It should still be a while before going into beta testing. Production and Commercialisation will be handled by a company in Portugal, since this is where I am living at the moment, even though I have a South African callsign. I will be changing to a Portu!
gu!
> ese callsign soon.
>
> Too all, thank you and kind regards,
> ---
> Fernando M. I. Carreiro
> mailto:fmi@carreiro.com
> callsign:ZS6BLF
>
>
>

76. Re: [HFSIG:2915] Re: Why so little

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 30 Oct 1998 22:46:15 +0000
To: hfsig@tapr.org
Message-ID: <363A41B7.1A19EB31@dircon.co.uk>
Rendered body type: text/plain

Anthony Martin wrote:
>
> I don't mind looking stupid. It demonstrated a point.
>

You will find it in the tapr ftp archives, I believe it is
called newqpsk. From memory it is a 15 tone DQPSK modem. Johan
KC7WW wrote a bit about it in the TAPR newsletter a year or so ago.

I have not used it but I understand it works quite well on TCP/IP.
I am sure someone that has used it can enlighten us more. You need an
EVM to use it. But I think someone could do something similar using
a sound card.

Anyway your are certainly not stupid Anthony.

> OK Eduardo, so where do I get it? I'm using Sun Unix, so I don't
> frequent the linux places.

Well the distribution I use comes from a company called Red Hat. The
software is free and you can download it from the Internet. It sometimes
appears on the front of Computer Magazines. Also yesterday while I
was in the local bookshop I saw a book by John Wiley that not only
described the inner workings of the Kernel but has a Red Hat distribution
stuck on the back.

You could try the computer section of Clarkes in Chelmsford or one of the
various computer shops.

Finally the Win95 stuff is distributed at

http://www.forthnet.gr/sv2agw/start.htm

The part that makes packet radio look like a telephone modem
costs money, although the free!? version will run just about long enough
for you to try it out. I managed to get it to work ok.

- Charles

77. Re: [HFSIG:2915] Re: Why so little

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 30 Oct 1998 22:53:43 +0000
To: "hfsig@tapr.org" <hfsig@tapr.org>

Message-ID: <363A4377.745988C9@dircon.co.uk>
Rendered body type: text/plain

Charles Brain wrote:
>
> Well the distribution I use comes from a company called Red Hat. The
> software is free and you can download it from the Internet.

Linux is free, however the Red Hat distribution is about $20 dollars.
Linux includes X windows as well so you can play workstations! The
Red Hat version has both the free X-windows and a commercial one. So
it has a bit more than normal Linux. Well it does on the version I
have.

- Charles

78. Re: [HFSIG:2915] Re: Why so little

From: Eduardo Jacob <edu@kender.es>
Date: Sat, 31 Oct 1998 01:06:17 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19981031010617.00836c30@kender.es>
Rendered body type: text/plain

>OK Eduardo, so where do I get it? I'm using Sun Unix, so I don't
>frequent the linux places.

I use linux because neither me nor my employer can purchase a Sparc or
Silicon Graphics for my use...

>I've got a freebie SuSE Linux 5.2 CD in my briefcase.
>Is it on there somewhere? Oh no, I don't believe this. I've
>been carrying it around and yet I didn't know I had it.

You can get the kernel sources (sorry, not executable) if they are not on
that CD (it is very usual to ship the kernel source on the CD for compiling
a tailored kernel) directly from the author at:

The latest stable kernel at
ftp://ftp.funet.fi/pub/Linux/PEOPLE/Linus/v2.0

The latest development kernel (will soon be 2.2 stable version) at
ftp://ftp.funet.fi/pub/Linux/PEOPLE/Linus/v2.1

Beware, several Mb compressed and perhaps about 15-20Mbs uncompressed...

You can also take a look at Tom Sailer page at:

http://www.ife.ee.ethz.ch/~sailer/ham/ham.html

He has many modems available for the SB there, including packet modems,
rtty, amtor ,pactor 1 and HF fax.

There is also psk31 is available for Linux and SB.

See http://wwwcip.informatik.uni-erlangen.de/user/hsreiser/hamradio/#psk

>So enlighten me chaps, which of the HF modem implementations
>mentioned on hfsig will actually work with this sweet linux
>AX-25 stack :-)

Any one that offers a kiss interface like newqpsk.

In any case please take the time to take a look at the Linux and HAM FAQ
it doesn't show latest advances but perhaps could convince to take a look
at it and perhaps you will be surprised.

http://sunsite.unc.edu/mdw/HOWTO/HAM-HOWTO.html

Regards
Eduardo/EA2BAJ

79. Re: [HFSIG:2918] Re: Why so little

From: Bob Johnson <bobj@cisi.com>
Date: Fri, 30 Oct 1998 19:18:09 -0500
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19981030191809.00761ce0@mail.atlantic.net>
Rendered body type: text/plain

>> Well the distribution I use comes from a company called Red Hat. The
>> software is free and you can download it from the Internet.
>
>Linux is free, however the Red Hat distribution is about $20 dollars.
[....]

Redhat Linux is available free at ftp://ftp.redhat.com

You might want to read the installation instructions at http://www.redhat.com

I believe the Linux distribution license requires that it be distributed
free. If you buy the boxed set from Redhat, you are paying them for
support that isn't available to those who download the free distribution.

- Bob Johnson
  WB4JCM

80. Re: Why so little participation?

From: "Fernando M. I. Carreiro" <fmi@carreiro.com>
Date: Sat, 31 Oct 1998 01:41:24 -0000
To: <hfsig@tapr.org>
Message-ID: <000401be046f$c90ff620$e6ffa8c0@pc001>
Rendered body type: text/plain

Hello again,

    > Where to find Linux AX.25 Source:

    Try the following FTP sites: "sunsite.unc.edu" and "ftp.dungeon.com" at directories "/pub/Linux" and
 "/pub/Linux/distributions". They have various flavours and versions of Linux - Red Hat, Slackware, etc.
 They have both binaries and full source code.

    > Walt DuBose wrote:
    > 1) What is the maximum baud rate (or phase shift)
    > 2) What is the signal bandwidth?

    Well it is not easy to answer these questions simply, as the VPSK mode is kind of uncommon. Although
 I am currently using VPSK for testing, it is or will soon be a patented or licensed mode. For that
reason I am in the process of testing a new variation of it so as not to conflict with the patent. The
variation should actually work better than the original VPSK. I will however explain the VPSK
implementation for now.

    I am using 6 tones or sub-carriers much like other popular HF modes (such as "newqpsk"). However,
they use VPSK instead of BPSK or QPSK. VPSK has a much narrower bandwidth and is much better for clock
recovery. Each tone operates at a different bit rate and bandwidth from the other tones. I do however
use pure sine waves instead of pulse shaped square waves as in VPSK. This gives a much cleaner signal.
In order to answer the above question I will describe one of the tones in this mode. The other tones
will operated at different values, but use the same concept. It is the sum of all six tones that gives
the high data rate.

    Example: For 5VPSK the total bit rate is 15150.86bps, with 6 tones in the range 456.824Hz to 2205Hz
(1748.176Hz bandwidth). Compression factor is 8.666 bits/sec/Hz. The highest tone sits between 2205Hz
and 1696.154Hz (508.846Hz bandwidth). The carrier tone for this slot is 2205Hz. The bit rate is 4410bps
and the BAUD rate is 3969Hz. There 3 levels of phase shift that occur every half wave. The half wave
phase shifts are 0deg, -36deg, -72deg. The shifts are applied such as to lower the frequency of the
tone. The sampling frequency required for this slot is 44100Hz (chosen so as to be compatible to sound
cards). The lowest tone for this mode sits between 593.871Hz and 456.824Hz (105.421Hz bandwidth). The
carrier tone for this slot is 593.871Hz. The bit rate is 913.647bps and the BAUD rate is 822.282Hz.

    Well, hope that has answered the questions. If not, let know what specifics you wish me to focus on?

Fernando M. I. Carreiro
mailto:fmi@carreiro.com
http://www.fmi.carreiro.com
callsign:ZS6BLF

81. HF digital activity is growing!

From: Marijan Miletic <s56a@s55tcp.ampr.org>
Date: Sat, 31 Oct 98 09:02:03 EST
To: hfsig@tapr.org
Message-ID: <34528@s55tcp.ampr.org>
Rendered body type: text/plain

Once upon a time there were some interesting experiments with new HF digital
modes by the members of this list.

Due to Internet, number of USA packet nodes has halved! We try to swim against
the currents by making our own 1.2 Mbits GHz links and home TNC but the

applications are not clear. It should certainly involve graphics...

While CW seems to be decaying, Baudot 45,45 wpm RTTY 170Hz FSK is very much
growing as witnessed in HF contests. Number of East EU participants follows
the availabilty of PC there. The only extra needed is a SoundBlaster which
costs peanuts. Excellent BTL freeware is widely available.

I guess higher speed multitone two-way digital modes on PC/SB would be embraced
by a wide ham community. Windows would be preffered over Linux :-)

73 de Mario, S56A, N1YU.

82. Re: [HFSIG:2922] HF digital activity is growing!

From: Eduardo Jacob <edu@kender.es>
Date: Sat, 31 Oct 1998 13:26:53 -0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19981031132653.00823810@kender.es>
Rendered body type: text/plain

>Windows would be preffered over Linux :-)

        Linux really cost peanuts.....

        Eduardo/EA2BAJ

83. Re: [HFSIG:2923] Re: HF digital activity is growing!

From: W6RCecilA <w6rca@ibm.net>
Date: Sat, 31 Oct 1998 08:10:08 -0700
To: hfsig@tapr.org
Message-ID: <363B2850.439@ibm.net>
Rendered body type: text/plain

Eduardo Jacob wrote:
>
> >Windows would be preffered over Linux :-)
>
>   Linux really cost peanuts.....

Linux is somewhat like CW. It costs almost nothing but
has a steep learning curve.
--
73, Cecil, W6RCA   http://people.delphi.com/CecilMoore

84. Re: [HFSIG:2922] HF digital activity is growing!

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Sat, 31 Oct 1998 12:06:43 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.4.05L.9810311155340.19509-100000@arctic>
Rendered body type: text/plain

On Sat, 31 Oct 1998, Marijan Miletic wrote:

> Due to Internet, number of USA packet nodes has halved!
> ... applications are not clear. It should certainly involve graphics...
>
> While CW seems to be decaying, Baudot 45,45 wpm RTTY 170Hz FSK is very much
> growing as witnessed in HF contests.

This might possible explain the phenominal growth of APRS. Not only do
you get to "see" everyone on the map, but it also shows you instantly
where propogation is open. We are now beginning to see Eruopeans pop up on
our APRS world maps coming in via the 10.149.2 APRS National tracking
channel. Due to there being 80 or more gateways between that frequency
and VHF (144.39 in the USA), then any APRS packet transmitted on that
frequency wll be seen by THOUSANDS of stations, potentially worldwide.

You can even check to see your own transmitted packet by monitoring the
worldwide APRS stream by telnetting to 199.227.86.221 port 23. The
APRServe internet stream is like the worlds largest space diversity HF
receive system. As long as ONE of the 80 or more APRS HF gateways hears
your packet, it will be seen throughout the network.

APRS packets on 10.149.2 (mark) are usually transmited using the UNPROTO
path of APRSHF VIA GATE,WIDE. THis way they go through the Gateways
and then are digipeated by the nearest APRS "WIDE" digipeater to any
listening APRServe IGATES.

Come Join us if you want to play with something new on HF packet that

takes NO new hardware...

de WB4APR, Bob

85. PSK31

From: "M. Greenman" <coombedn@ihug.co.nz>
Date: Sun, 1 Nov 1998 18:11:18 +1300
To: <hfsig@tapr.org>
Message-ID: <199811010512.SAA11648@smtp2.ihug.co.nz>
Rendered body type: text/plain

I'm looking for contacts to try out PSK31 - any offers?

In the light of all the complaints about inactivity - who's game to try
PSK31?

73,

Murray ZL1BPU

November 1998

Original source file: hfsig.9811
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9811

1. Re: [HFSIG:2925] Re: HF digital activity is growing!

From: Marijan Miletic <s56a@s55tcp.ampr.org>
Date: Sun, 01 Nov 98 12:17:03 EST
To: hfsig@tapr.org
Message-ID: <35068@s55tcp.ampr.org>
Rendered body type: text/plain

ARPS looks to me as another sysop's toy and I am not sure of HF appeal!

73 de Mario, S56A, N1YU just out of CW & RTTY contests!

---

2. Re: [HFSIG:2910] Re: Why so little participation?

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Mon, 02 Nov 1998 11:29:21 +0100
To: hfsig@tapr.org
Message-ID: <363D8981.56C71E6C@ife.ee.ethz.ch>
Rendered body type: text/plain

Anthony Martin wrote:

> I can't see how you would be able to get hold of the
> TCP/IP packets either without having to alter the
> operating system. Once they're formatted, if they're not
> addressed to yourself they inevitably go straight out
> of the machine on ethernet/slip/ppp!

The idea is to write kind of a complex "network adapter driver"
and install it into the OS. It automatically gets every
packet that is routed to the network interface. You use
the normal routing capabilities of the OS to determine
which packet has to go out on HF and which ones on
the ethernet.

If you don't want to hack kernel mode drivers, you
can quite easily (on Linux and on any other non braindead
Unix) attach a SLIP device over a pty/tty pair and attach
your code on the other side. This gives you essentially
a user mode network driver. You have to unpack
SLIP and remove the header, but this is peanuts at
"HF speeds".

> At the socket level, internet access is very simple.
>
> Anyone can write a simple client or server.
>
> Client applications normally access TCP/IP sockets
> via 5 main C calls:
> connect()   bind() read() write() close()
> On unix they may also use:
> select() or poll()
> A server may additionally use:
> accept()

Basically yes. But you'll have to use different
arguments to these calls, such as PF_HF for
socket, as well as a different struct sockaddr
for connect/bind.

The major problem is probably that applications
like web servers/proxies (apache) do access control
by IP address.

Tom

---

3. Re: [HFSIG:2919] Re: Why so little

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Mon, 02 Nov 1998 18:25:32 +0000
To: hfsig@tapr.org
Message-ID: <363DF91C.635C@marconicomms.com>
Rendered body type: text/plain

Eduardo Jacob wrote:
>

> >OK Eduardo, so where do I get it? I'm using Sun Unix, so I don't
> >frequent the linux places.
>
>   I use linux because neither me nor my employer can purchase a Sparc or
> Silicon Graphics for my use...
>
Solaris for x86 (ie. PCs) is available free from Sun for non-commercial
applications. There is a nomial fee for posting the CDs. It was last
time I looked.

SCO Unixware is also available free for non-commercial use.

I'm on a 3-year-old SparcStation 20. It happily supports 2 or 3 users, a
web
server and other bits'n'bobs.

At the time, the TI DSP tools that offered multiprocessor DSP emulation
&
debugging were only available under Unix. So we went Unix. Being a
workstation
the tools can be used from anywhere in the company. Up to 7 different
people have been using them yet we have 1 licence & maintenance
agreement so
I think we got our money's worth.

====

I don't like the way that the members of this group are polarised into
Linux and Windows users, and EVM haves & have-nots.

The arguaments of `this platform is best for hams' `there are more users
on that platform', are not constructive. People have their platforms
that they
like and which suit their particular needs and you won't change them.

There's no point pushing users down any platform route. They'll reject
you.

I think it's clear that a fair proportion of hams are committed Linux
users.

Which platform people have should not exclude people from communication
with some modulations and protocols whilst enabling them to do others.

====

I think we need to do an FAQ lising the protocols & modem
implementations
and on-air waveforms, their interoperability, platform compatability,
where to get them etc. The FAQ should cover the `how-to' of getting
on air with HF modems and HF protocols.

(Has that been done too?)

An FAQ would highlight the incompatabilities between modems and
protocols and platforms where ports don't exist.

====

I'm going to pick up the AX-25 Linux code to see what I can do with it.

And I'll look out for info on the KISS interface.

Ant
--
Recovering from `changed corporate identity syndrome'
_____

A.N.Martin   anthony.n.martin@marconicomms.com   www.marconicomms.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
Marconi Communications UK, Marconi House, New St. Chelmsford, UK
_____

4. Meteor scatter

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 02 Nov 1998 20:40:35 -0800
To: hfsig@tapr.org

Message-ID: <199811030444.UAA27096@peak.org>
Rendered body type: text/plain

Hi all,

Does anyone perhaps have ideas suitable for
doing meteor scatter communications?

The main challenges are inconsistent and weak signals with
pretty hefty doppler effects when there is any signal. What
would be suitable for modulation and data format?

Thanks in advance,

Johan, KC7WW

5. Re: [HFSIG:2929] Re: Why so little

From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
Date: Tue, 03 Nov 1998 10:20:37 +0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19981103102037.00893c00@bicc00.bi.ehu.es>
Rendered body type: text/plain

>>   I use linux because neither me nor my employer can purchase a
Sparc >> or Silicon Graphics for my use...

        Well, I should have say the increase in my productivity perhaps would not
cover the increased cost of the hw/sw :-)

        When I think of those workstations, I think in more computing power,
integrated audio/video capture and a big screen with much resolution. It's
certainly not a matter of available software...

>Solaris for x86 (ie. PCs) is available free from Sun for non-commercial
>applications. There is a nomial fee for posting the CDs. It was last
>time I looked. SCO Unixware is also available free for non-commercial use.

        Yes, I knew of them. But I do think that linux offers much more than those.

>There's no point pushing users down any platform route. They'll reject
>you.

        Before of all, at the present time I almost don't program. And if I do
it's at a very high level, not the bit-level. Years ago I programmed at
bit-level in C and several assemblers, but I have never thougth of myself
as a brilliant programmer. Nowadays I install (compile, modify, tweak) and
use many public/free applications in many languages on linux and other
unixes, and install and run many applications on windows and NT.

        I only would like to say that acording to my experience, linux/unix
software generally have some properties that are interesting to the ham
community:
        - Full source code: That let's an open door.
        - More portability
        - More itself compatibility ...

        Speaking of open doors: Think of the AEA DSP2232, on the paper it was an
open platform based on a 56001 and much software was expected. Now people
is selling them to purchase EVM56002 because a development package was
never made available and there is almost no (I don't know of it) additional
software for it. Probably, many things could have been done for the AEA
that would result in benefits for new HF modes: PSK31 probably could be
implemented, based on the implementations for the EVM56002 from Peter G3PLX.

        The point for me is that if somebody is willing to learn programming for
designing ham applications/modems, I think it's work and effort will be
more valuable if it is linux/unix oriented. If you want to include a
commercial benefit, this is probably not true.

>I think it's clear that a fair proportion of hams are committed Linux
>users.

        So do I.

>I think we need to do an FAQ lising the protocols & modem
>implementations
>and on-air waveforms, their interoperability, platform compatability,
>where to get them etc. The FAQ should cover the `how-to' of getting
>on air with HF modems and HF protocols.

Yes. That would be very fine...

>And I'll look out for info on the KISS interface.

It's here:

http://people.qualcomm.com/karn/papers/kiss.html

Regards

Eduardo/EA2BAJ

6. Kiss

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Tue, 03 Nov 1998 16:14:28 +0000
To: hfsig@tapr.org
Message-ID: <363F2BE4.34BB@marconicomms.com>
Rendered body type: text/plain

> http://people.qualcomm.com/karn/papers/kiss.html

Thanks for that URL Eduardo.

Kiss is quite unlike the usual HF modem DCE interfaces.

A few questions about KISS...

Is the use of HDLC mandatory for AX25? HDLC gets you
byte-sync and packet-start information from a bitstream.
If an modem already has a means for doing this, HDLC
would appear to be unnecessary.

Is it necessary to clear down the transmitter between packets
and check the channel is free, or do they follow consecutively
within the same transmission?

Where thew modem is tightly integrated with the transceiver,
TXDELAY and TXtail aren't necessary. Any variable tune delays
of the tranceiver are handled by arrangements between the
modem & transceiver.

The SlotTime interval units of 10ms are too fine. HF sync
sequences are of the order of 600ms, and link delays of
the order of 5ms. This suggests much coarser units should be
used. If you stick with 10ms collisions will be serious.

It may be only practical to use 6 slots about 300ms apart.

When implementing KISS on a DSP based HF modem the missing
Tx flow control is a nuisance. Something like 32k of RAM is
required for a packet buffer. Tx flow control would mean
that the modem would work fine on 4k. This would allow
implementation in a 'C50 DSK. It would also free that
DSP RAM you need for better things like interleave,
equalisation & FEC.

I note KISS does not offer any scanning/frequency selection
facilities; it's only intended for single-channel operation.

I'm thinking about adding support for KISS on an existing
modem so I can try out AX-25.

--
Recovering from `changed corporate identity syndrome'

A.N.Martin anthony.n.martin@marconicomms.com www.marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
Marconi Communications UK, Marconi House, New St. Chelmsford, UK

7. Re: [HFSIG:2932] Kiss

From: Lyle Johnson <lylej@azstarnet.com>
Date: Tue, 03 Nov 1998 16:59:21 -0700
To: hfsig@tapr.org

Message-ID: <3.0.5.32.19981103165921.008d5d10@pop.azstarnet.com>
Rendered body type: text/plain

>A few questions about KISS...
>
>Is the use of HDLC mandatory for AX25?

It is part of the AX25 Level 2 protocol spec.

>Is it necessary to clear down the transmitter between packets
>and check the channel is free, or do they follow consecutively
>within the same transmission?

Once the chanel is yours, you can use it for multiple frames. The limit on
frames is 7 due to the modulo-8 counter used for ACKs. If the channel is
noisy, it's best to send just oen frame. If the channel is very good, you
may safely send more. There is no slective ACK, it is "go back N", so if
in doubt, just send one frame.

>Where thew modem is tightly integrated with the transceiver,
>TXDELAY and TXtail aren't necessary. Any variable tune delays
>of the tranceiver are handled by arrangements between the
>modem & transceiver.

This is part of KISS and not AX25. But TXDELAY also needs to tak einto
account the time it takes the REMOTE receiver to sync to your data stream
and prepare for your opening flag. Bext so send a stream of ZERO bits
(before NRZI encoding) to maximize transition density.

>The SlotTime interval units of 10ms are too fine. HF sync
>sequences are of the order of 600ms, and link delays of
>the order of 5ms. This suggests much coarser units should be
>used. If you stick with 10ms collisions will be serious.

Again, this is not part of AX25. The granularity is more related to local
Tx key-up DETECTION by the remote unit, not frame size.

>When implementing KISS on a DSP based HF modem the missing
>Tx flow control is a nuisance. Something like 32k of RAM is
>required for a packet buffer. Tx flow control would mean
>that the modem would work fine on 4k. This would allow
>implementation in a 'C50 DSK. It would also free that
>DSP RAM you need for better things like interleave,
>equalisation & FEC.

This has always been a weakness of KISS. SOme europeans have implemented
some enhancements to this, I believe, which incldues flow control.

>I note KISS does not offer any scanning/frequency selection
>facilities; it's only intended for single-channel operation.

It is intended to deal with data streams, not radio channels.

Enjoy!

Lyle

8. Re: HF digital activity is growing!

From: k5cnf1@juno.com (Richard B Morrow)
Date: Tue, 3 Nov 1998 20:52:34 -0600
To: hfsig@tapr.org
Message-ID: <19981103.205234.4582.5.k5cnf1@juno.com>
Rendered body type: text/plain

What you guys need to keep in mind is that whatever you come up with,
MUST work with what equipment people have with very little interfacing
problems or it will not fly with the multitudes. Also so many hams now
are NOT techincally inclined now.

Richard, K5CNF

_____
You don't need to buy Internet access to use free Internet e-mail.
Get completely free e-mail from Juno at http://www.juno.com/getjuno.html
or call Juno at (800) 654-JUNO [654-5866]

9. Soundcard

From: malberts@ncs.com.na (Mike Alberts)
Date: Wed, 04 Nov 1998 11:05:55 +0200

To: HF mail <hfsig@tapr.org>
Message-ID: <364018F3.A4AE7739@ncs.com.na>
Rendered body type: text/plain

Hi All.

Could anyone advise me on setting up a PCI 64 soundcard to use in the
HAM shack....I am sure it has features I can utilise.
Thanks
Mike V51MA
malberts@ncs.com.na

10. Re: Kiss

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Wed, 04 Nov 1998 12:14:46 +0100
To: hfsig@tapr.org
Message-ID: <36403726.7E3EE97F@ife.ee.ethz.ch>
Rendered body type: text/plain

Anthony Martin wrote:

> Is the use of HDLC mandatory for AX25? HDLC gets you
> byte-sync and packet-start information from a bitstream.
> If an modem already has a means for doing this, HDLC
> would appear to be unnecessary.

HDLC also gives you a (weak) CRC, which you'd need to
supply a replacement for it. There are no other
means to detect transmission errors.

> Where thew modem is tightly integrated with the transceiver,
> TXDELAY and TXtail aren't necessary. Any variable tune delays
> of the tranceiver are handled by arrangements between the
> modem & transceiver.

This is all part of the channel access algorithm, which
is often on VHF amateur packet radio p-persistent CSMA.
But this choice is not always adequate. In these environments,
KISS is really a bad choice. Also, KISS gives you no feedback
on when a queued packet in a TNC is actually transmitted
over the air, so the AX.25 stack has no chance to start
a retransmission timer when the packet actually appears
on the air. Rather, it starts the timer when the packet
is queued to the TNC, and risks unnecessary retransmissions.
Thus, on a busy channel, KISS makes things worse than
they already are.

6PACK, as an alternative to KISS, couples protocol
stack and TNC much tighter and also moves the channel
access algorithm away from the TNC to the PC, but
this is probably not what you want for your application.

> I note KISS does not offer any scanning/frequency selection
> facilities; it's only intended for single-channel operation.

It can do multiple channels; but indeed there are no
standard protocol elements to set modem parameters. However,
this seems to be difficult to do generically, as every
modem has different parameters...

Tom

11. Re: [HFSIG:2936] Re: Kiss

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Wed, 04 Nov 1998 18:01:10 +0000
To: hfsig@tapr.org
Message-ID: <36409666.41C4@marconicomms.com>
Rendered body type: text/plain

Thomas Sailer wrote:
>
> Anthony Martin wrote:
>
> > Is the use of HDLC mandatory for AX25? HDLC gets you
> > byte-sync and packet-start information from a bitstream.
> > If an modem already has a means for doing this, HDLC
> > would appear to be unnecessary.
>
> HDLC also gives you a (weak) CRC, which you'd need to

> supply a replacement for it. There are no other
> means to detect transmission errors.
>
How does the AX-25 stack know there's been a transmission
error if the KISS TNC handles the HDLC CRC?

I had assumed that the TNC handled all the HDLC, decoding
the HDLC, removing flags and the checksum and passing the
raw data up via KISS to AX-25.

Transmit packets received from the KISS interface are raw
data to be HDLC encoded, and then have flags and CRC added.

Have I got that right?

If the TNC checks the CRC, what does it do with the
result? does it silently dump received packets that
are in error?

I think I'll have to get the AX-25 code and an AX-25
documentation to make sense of KISS, the text is not
very rigorous by itself.

I'm still looking for a Linux site where I can get one
big tarfile, not 50 little ones.

I agree with your comments on the retransmit timer;
KISS breaks AX-25 as on busy channels, it just shovels
in more repeat packets. Bad news.

> It can do multiple channels; but indeed there are no
> standard protocol elements to set modem parameters. However,
> this seems to be difficult to do generically, as every
> modem has different parameters...

You have the implementation-specific `SetHardware'
command which should enable the choice of up to 256
combinations of mode, bitrate etc.

If the modem is capable of determining the incoming baudrate
there's no way of reporting that to the stack so that the
modem can be set to reply in kind. I think it should be
allowed to report appropriate values of `SetHardware'.

I think there should be some kind of basic echo response
to verify your modem is switched on and connected to the
computer properly!

> Tom

--
Recovering from `changed corporate identity syndrome'
____

A.N.Martin anthony.n.martin@marconicomms.com www.marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
Marconi Communications UK, Marconi House, New St. Chelmsford, UK
____

12. Re: [HFSIG:2934] Re: HF digital activity is growing!

From: "Harry L. Styron" <hlstyron@wcrklaw.com>
Date: Wed, 04 Nov 1998 10:58:08 -0800
To: hfsig@tapr.org
Message-ID: <3640A3C0.68E2DE50@wcrklaw.com>
Rendered body type: text/plain

Richard B Morrow wrote:
>
> What you guys need to keep in mind is that whatever you come up with,
> MUST work with what equipment people have with very little interfacing
> problems or it will not fly with the multitudes. Also so many hams now
> are NOT techincally inclined now.
>
With all due respect, limiting innovation by requiring it to be within
what the masses can presently understand and use is, by definition,
killing it. After all, how many users understand what is going on in a
56K telephone modem? When we get to an OSI level 0 protocol which is

truly superior to what we have now, it will be put into a black box and
made available to everyone. "If you build it, they will come!"

Harry Styron, K6MFV

13. Re: [HFSIG:2938] Re: HF digital activity is growing!

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Wed, 04 Nov 1998 21:59:24 +0000
To: hfsig@tapr.org
Message-ID: <3640CE3C.4234@marconicomms.com>
Rendered body type: text/plain

Harry L. Styron wrote:
>
> Richard B Morrow wrote:
> >
> > What you guys need to keep in mind is that whatever you come up with,
> > MUST work with what equipment people have with very little interfacing
> > problems or it will not fly with the multitudes. Also so many hams now
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> 56K telephone modem? When we get to an OSI level 0 protocol which is
> truly superior to what we have now, it will be put into a black box and
> made available to everyone. "If you build it, they will come!"
>

How are you going to interest anyone in a technical
subject like ham radio if you put things in a box and
hide everything technical?

How do you interest anyone in art if you put it in a
box and hide the technical stuff like colours (colors)
shapes and texture?

Today people are expected to have lots of general knowledge
about history, current affairs, politics, medicine, film,
music & art. Yet people somehow think technology and
engineering are different. Being nuts 'n bolts and guts of
things that make things work they think it's something
they shouldn't have to know about. And to avoid embarassing
anyone you should avoid talking about it at parties. Yet
Golf of Skiing is fine. And the more important someone
is the less they should know about it.

On the TV we get programmes about complicated technical
subjects like the AIDs virus, carreers of pop musicians,
european union agricultural policy, D-Day landings in the
2nd world war, important people inthe fashion world,
and the life of a lemming colony.

When do we ever see programs on car mechanics, programming,
Microprocessor architecture, radio receiver design,
HF propagation, and the IP protocol, the work of CERN...

Yep I agree that it has to be black-boxed for the masses
to have it. And very sorry for the poor state of the
masses I am too. Many can't operate their VCR.

The masses need educating. Ham radio should not sell out.
If the masses are in a decrepit state it doesn't mean
that ham radio should follow suit. Describing hams as
`not technically inclined' should be a gross insult.

If Ham radio is about black boxes, why would anyone
bother when they can use a mobile phone to call the
other side of the planet? You don't need to be a ham
to use a box. If you make ham radio about box users,
you'll kill it.

Is it really just about needing a licence to use a
black box with a certain label on? (Technical terms
like `wavelength' and `erp' aren't used by most hams)
It's then only a small step to abolishing the licence
and leaving the black boxes to compete for spectrum
on a commercial basis.

There are a very large number of technically inclined
people who might become hams if encouraged,
particularly as technical skills aren't highly valued
elsewhere, and if they did it would be much better
for ham radio.

I don't think hams should expect `a few good men' who
are technical to come up with the goods, and then
expect it to be given away at the lowest possible
cost. All hams should pull their weight in advancing
some aspect of the hobby.

We should continue to prize technical competence
in hams and expect it of hams.

OTOH Linux AX-25 is way way too complicated, it beats
me....

Anyone got
"Mickey Mouse's guide to Linux AX-25 for someone who's
never used AX-25 before and knows not ye legends of
ye olde TNC" appendix A "Alternatives for users that
can't recompile the Unix kernel" :-)

Most unix software is portable. This ain't. It's
been bolted down and concreted in. Permanently.

Linux AX-25 looks good though. Complete implementation
of every protocol you could ever want including TCP/IP
right down to modems for your soundcard... Nothing to
buy. BBS, servers, shell logins, they're are all there.

If I could cross-compile linux maybe I can get it to
fit on an old 386...

--
Recovering from `changed corporate identity syndrome'

A.N.Martin anthony.n.martin@marconicomms.com www.marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
Marconi Communications UK, Marconi House, New St. Chelmsford, UK

14. Re: HF digital activity is growing!

From: k5cnf1@juno.com (Richard B Morrow)
Date: Wed, 4 Nov 1998 20:56:03 -0600
To: hfsig@tapr.org
Message-ID: <19981104.211110.4582.12.k5cnf1@juno.com>
Rendered body type: text/plain

>> What you guys need to keep in mind is that whatever you come up with,
MUST work with what equipment people have with very little interfacing
>> problems or it will not fly with the multitudes. Also so many hams now
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killing it. After all, how many users understand what is going on in
>a 56K telephone modem? When we get to an OSI level 0 protocol which is
truly superior to what we have now, it will be put into a black box and
made available to everyone. "If you build it, they will come!"
>
Harry, I have had my call for 43 years and a commercial (1st class phone
with ship radar) for 41 years and what I hear on the bands is a definate
lack of smarts and abililty that is increasing with time. One local guy
asked me why I would want an ancient Johnson Thunderbolt, since it had
to be tuned everytime you change bands. Not hard to understand when you
look at the tests now. My first ham tests were real tests, with draw the
diagrams and label all parts. THe commercial tests that I took made a
stack of papers 3 inches tall everytime I took one. Now the same tests
are made up of no more than 100 questions.

Look at what Windows did for computers. made it easy for the masses to

use them ease, certainly more easy than a TRS-80 Model 1 and TRSDOS. Been there and done that.

CUL, RM K5CNF

_____
You don't need to buy Internet access to use free Internet e-mail.
Get completely free e-mail from Juno at http://www.juno.com/getjuno.html
or call Juno at (800) 654-JUNO [654-5866]

15. Re: HF digital activity is growing!

From: k5cnf1@juno.com (Richard B Morrow)
Date: Wed, 4 Nov 1998 21:11:10 -0600
To: hfsig@tapr.org
Message-ID: <19981104.211110.4582.13.k5cnf1@juno.com>
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>subject like ham radio if you put things in a box and
>hide everything technical?
>
It will not be easy but there will be a few who will become interested in our hobby when they find out it is a challenge.

>How do you interest anyone in art if you put it in a
>box and hide the technical stuff like colours (colors)
>shapes and texture?
>
I do not think that 60% or more of the ham population really care about the tech stuff, as long as it works.
>
>Today people are expected to have lots of general knowledge about
history, current affairs, politics, medicine, film, music & art. Yet
people somehow think technology and engineering are different. Being nuts
'n bolts and guts of things that make things work they think it's
something they shouldn't have to know about. And to avoid embarassing
anyone you should avoid talking about it at parties. Yet Golf of Skiing
is fine. And the more important someone is the less they should know
about it.
>
That has been the case for as long as I can remember it. Science has been a nerd and geek subject for as long as I can remember. Particularly if you are a teenager. But computers are not really considered geeky. Beats the hell out of me why.

>On the TV we get programmes about complicated technical subjects like
the AIDs virus, carreers of pop musicians, european union agricultural
policy, D-Day landings in the 2nd world war, important people inthe
fashion world,and the life of a lemming colony.
>
>When do we ever see programs on car mechanics, programming,
Microprocessor architecture, radio receiver design, HF propagation, and
the IP protocol, the work of CERN... >

There are some here, and I am always watching them when they come on. but speaking from experince, Getting people to try new stuff can be damn hard. Most of the stuff I watch is on the Discovery channel and similar labeled channels.

>Yep I agree that it has to be black-boxed for the masses
>to have it. And very sorry for the poor state of the
>masses I am too. Many can't operate their VCR.
>
>The masses need educating. Ham radio should not sell out.If the masses
are in a decrepit state it doesn't mean that ham radio should follow
suit. Describing hams as`not technically inclined' should be a gross
insult.
>
But it is true in many cases. Far more than 40 years ago.

>If Ham radio is about black boxes, why would anyone bother when they can
use a mobile phone to call the other side of the planet? You don't need
to be a ham to use a box. If you make ham radio about box users,you'll
kill it.
>
I am in agreement with what you said, but a lot of the hams here are EX CB types and a box is all that they want

>Is it really just about needing a licence to use a

>black box with a certain label on? (Technical terms
>like `wavelength' and `erp' aren't used by most hams) It's then only a
small step to abolishing the licence and leaving the black boxes to
compete for spectrum on a commercial basis.
>
>There are a very large number of technically inclined people who might
become hams if encouraged, particularly as technical skills aren't highly
valued elsewhere, and if they did it would be much better for ham radio.

That is true.

>I don't think hams should expect `a few good men' who are technical to
come up with the goods, and then expect it to be given away at the
lowest possible cost. All hams should pull their weight in advancing some
aspect of the hobby.
>
You are right.

>We should continue to prize technical competence
>in hams and expect it of hams.
>
That is definately true.
>
>OTOH Linux AX-25 is way way too complicated, it beats
>me....
>
I hope it is not contageous.

CUL, RM, K5CNF

_____
You don't need to buy Internet access to use free Internet e-mail.
Get completely free e-mail from Juno at http://www.juno.com/getjuno.html
or call Juno at (800) 654-JUNO [654-5866]

16. Re: [HFSIG:2937] Re: Kiss

From: Alexander Kurpiers <kurpiers@zeus.uet.e-technik.tu-darmstadt.de>
Date: Thu, 05 Nov 1998 12:17:03 +0100
To: hfsig@tapr.org
Message-ID: <3641892F.CDE2E367@zeus.uet.e-technik.tu-darmstadt.de>
Rendered body type: text/plain

Anthony Martin wrote:
>
> Thomas Sailer wrote:
> >
> > Anthony Martin wrote:
> >
> > > Is the use of HDLC mandatory for AX25? HDLC gets you
> > > byte-sync and packet-start information from a bitstream.
> > > If an modem already has a means for doing this, HDLC
> > > would appear to be unnecessary.
> >
> > HDLC also gives you a (weak) CRC, which you'd need to
> > supply a replacement for it. There are no other
> > means to detect transmission errors.
> >
> How does the AX-25 stack know there's been a transmission
> error if the KISS TNC handles the HDLC CRC?

You have to implement a timeout as the address field which carries
the information about the packet is protected with the same CRC that
protects the data part. So if the CRC is wrong, the whole packet
is discarded in the KISS TNC. The only way to find out, if a packet
is not received correctly is to *ask* if an acknowledge frame for this
information packet was missed or if the information packet itself
was not received with correct CRC.
HDLC and X25 are not very well suited to be used on wireless channels...

>
> I had assumed that the TNC handled all the HDLC, decoding
> the HDLC, removing flags and the checksum and passing the
> raw data up via KISS to AX-25.
>
> Transmit packets received from the KISS interface are raw
> data to be HDLC encoded, and then have flags and CRC added.
>
> Have I got that right?

Yes.

>
> If the TNC checks the CRC, what does it do with the
> result? does it silently dump received packets that
> are in error?

Yes, there is no way to correct the packets with wrong CRC (or
at least - I wouldn´t do it with just 16 bit CRC, too little
redundandy).
So throw it away...

>
> I think I'll have to get the AX-25 code and an AX-25
> documentation to make sense of KISS, the text is not
> very rigorous by itself.
>
> I'm still looking for a Linux site where I can get one
> big tarfile, not 50 little ones.

The AX25 implementation in the linux kernel is really not too good
currently.
I think the better way would be to look into the AX25 specifications.
They
should be somewhere on www.tapr.org.

Guide a good AX25 implementation is PC/FlexNet
(www.afthd.tu-darmstadt.de/~flexnet).
It runs under W95 or DOS and handles already the adaption between TCP/IP
and
AX25 (i.e. removing retransmissions from the TCP layer if you are using
an AX25
virtual circuit). PC/FlexNet has a API for low level drivers and
applications, but
no source code is available. It is free for non-commercial use in HAM
radio.

BTW I still doubt that AX25 is a good protocol to use it together with
HF modems.

73´ Alexander

--
-------------------------------+---------------------------------------
| Alexander F. Kurpiers |
|
| Institut fuer Uebertragungs- | Voice: +49-6151-162369
|
| technik u. Elektroakustik | Fax : +49-6151-165545
|
| Merckstrasse 25 | EMail: a.kurpiers@uet.tu-darmstadt.de
|
| D-64283 Darmstadt (Germany) | Hamradio: dl8aau@db0zdf.#rpl.deu.eu
|
-------------------------------+---------------------------------------

17. Bad Packets

From: Brian Beezley <k6sti@n2.net>
Date: Thu, 5 Nov 1998 04:47:44 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19981105044742.08573256@mail.n2.net>
Rendered body type: text/plain

>Yes, there is no way to correct the packets with wrong CRC (or
>at least - I wouldn´t do it with just 16 bit CRC, too little
>redundandy).
>So throw it away...

I regularly correct bad PACTOR packets by flipping each data bit in turn and
rechecking for valid 16-bit CRC. Since most errors are single-bit errors,
this frequently recovers unusable packets that would otherwise require
retransmission. I wouldn't use this method to recover critical data but for
ordinary conversational text it is plenty reliable.

Brian

18. Leonids Meteor Shower APRS TEST

From: Bob Bruninga <bruninga@nadn.navy.mil>
Date: Thu, 5 Nov 1998 09:27:25 -0500 (EST)
To: hfsig@tapr.org
Message-ID: <Pine.GSO.4.05L.9811050926070.5485-100000@arctic>
Rendered body type: text/plain

Although 6m and 10m are better, 2m may be "easier" for more people:

The Leonids Meteor Shower in two weeks (17-18 Nov) will be an excellent opportunity to experiment with MS on 2 meters using very brief Grid Square packets. During the Persieds in 1995, APRS stations recorded over 18 DX packets on 2m of paths between 600 to 1200 miles. This year we can do much better for two reasons.

1) There are far more APRS stations on the air
2) Since all of APRS has QSY'ed to 144.39 in most areas of the country, this leaves 145.79 available for a QRM free Meteor shower 2m packet frequency!

PACKET FORMATS: The best packet is one of the APRS formats so that other statons will instantly plot you on their maps. All packets should be transmitted as UI packets. Here are several APRS formats using my station and my gridsquare of FM19DX as an example.

WB4APR>FM19DX:]-[Bob		This is the Grid-In-TO format XMTed by APRSdos in MS mode. It will be plotted by all other APRS stations and all versions of APRS software
	TNC-ONLY:	This grid-in-to format can be sent by any non-Kantronics TNC by setting the UNPROTO to your gridsquare and setting BText to]-[xxxx... Then set B E 1... for 10 second XMT rate.
WB4APR>BEACON:[FM19DX]Bob		This format can be placed in Kantronics TNC BText and sent every 1 minute with B E 1. It is longer, and infrequent , but it will be decoded by all stations.
WB4APR>FM19DX:		This is the same as Grid-in-to and can be sent by any TNC by setting UNPROTO to your GRIDSQ and then going to CONVERSE and hitting ENTER key as often as you like! See timing below: It is the shortest possible position packet!
	NOTE:	THis format will NOT AUTOMATICALLY BE RECEIVED by other APRS stations unless they are in the Meteor scatter or GRID-IN-TO mode or have their FILTERS set to receive all packets.

APRSdos alt-S-FORMATS-MSCATTER MODE: This mode configures your station to receive GRID-IN-TO packets and to TRANSMIT them for 15 seconds continuous according to the regional timing plan. If you have less than 50 watts and dont want to QRM other locals, do not set APRSdos to MS format but set alt-S-FORMAT to grid-in-to. This way you can reveive MS packets. Both of these automatically set your FILTER to OTHER...

PATHS: Do not set your UNPROTO to anything. We dont want to see any digipeaters here!

TIMING: To organize transmitting an reception periods for this test, the USA is divided into 4 regions. Each will transmit for 15 seconds and then listen for the remaining 45 seconds Starting with the Northeast at 00 seconds followed by the SEast at 15 seconds and then the SWest at 30 seconds and then the NWest at 45 seconds of each minute. All of this timing is built into APRSdos if you are running alt-S-FORMATS-MScatter mode.

ICONS: The]-[is the HOUSE Icon. Use]Y[if you are using a Yagii, or]`[if using a dish, or any of the other hundred APRS icons...

BEAMING: Point your beam at one of the other 3 quadrants and go for it!

STATION REQUIREMENTS: Although some MS packets have been seen on OMNI antennas, these are very rare. Usually about 10 dB gain from a beam is required to receive these weak packets. Transmitting stations should run as much power as possible. ANything less than 100 watts will have to be very lucky.

FREQUENCY: All stations participating should operate on 145.79 except in
SOuthern California where 145.79 is still the main APRS channel. So here
are the recommendations:

   ALL USA (not SoCal): Operate MS on 145.79 for the duration of the
                         event. You may even hear SoCal regulars...

   Southern California: 1) Try to listen on 145.79 for weak MS packets
                            amidst the local QRM. Dont transmit 15 sec
                            key-down on 145.79 or you will make enemies...
                         2) Transmit high power on 144.39 and hope that
                            someone in the rest of the USA will hear you.
                            (But only accidentally, since the serious
                            MS stations will all be over on 145.79).
                            BUT you will make BIG enemies locally of the
                            ATV people that use 144.39
                         3) Listen on 144.39 for weak DX packets from all
                            the rest of the USA on APRS that are NOT
                            participating in MS on 145.79

If you run APRS, see the METEORS.TXT file included in all the README
files for more details. TEST beforehand. The APRSdos MScatter mode has
not been used much in the last 3 years, so I hope it still works!

de WB4APR, Bob

19. Re: [HFSIG:2937] Re: Kiss

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Thu, 05 Nov 1998 15:31:18 +0100
To: hfsig@tapr.org
Message-ID: <3641B6B6.EA549D80@ife.ee.ethz.ch>
Rendered body type: text/plain

Anthony Martin wrote:

> How does the AX-25 stack know there's been a transmission
> error if the KISS TNC handles the HDLC CRC?

The TNC just drops the corrupted packet, and the host never
notices the packet. It does not detect transmission
errors over the RS232, which can be a problem.

> Have I got that right?

Yes.

> If the TNC checks the CRC, what does it do with the
> result? does it silently dump received packets that
> are in error?

Yes. Sorry for the confusion, I just wanted to
say that your HF modem has to make sure the received
data is correct, as nobody else will do that.

Tom

20. Re: HF digital activity is growing!

From: "Wheatley, Maurice" <mwheatle@trsystems.com>
Date: Thu, 5 Nov 1998 09:31:17 -0500
To: "'hfsig'" <hfsig@tapr.org>
Message-ID: <93C4EE270865D21185BF0000F80163EA0160BD@WEBSERVER2>
Rendered body type: text/plain

>>If Ham radio is about black boxes, why would anyone
>>bother when they can use a mobile phone to call the
>>other side of the planet? You don't need to be a ham
>>to use a box. If you make ham radio about box users,
>>you'll kill it.

Just because someone doesn't care to understand how something works in
detail doesn't detract from their ability to enjoy using it and doesn't mean
they are technically incompetent. Some hams have indepth knowledge of EMF
theory and antenna design but don't know and could care less about how their
transceiver works. Do they enjoy the hobby any less?
Most hardware and software engineers use "black boxes" all the time. They
are called IC's. How many people design with these black boxes and don't
know and could care less how many mask layers and what ion implantation
processes were used to make the chip. It's all in where in the food chain

you like to be. Ham radio has many levels of activity from public service
to contesting to rag chewing to technical experimenting etc. Pick what you
like and enjoy it.

Moe
ae4jy

21. Re: [HFSIG:2943] Bad Packets

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Thu, 05 Nov 1998 15:37:40 +0100
To: hfsig@tapr.org
Message-ID: <3641B834.B0EBEB09@ife.ee.ethz.ch>
Rendered body type: text/plain

Brian Beezley wrote:

> I regularly correct bad PACTOR packets by flipping each data bit in turn and
> rechecking for valid 16-bit CRC. Since most errors are single-bit errors,
> this frequently recovers unusable packets that would otherwise require
> retransmission. I wouldn't use this method to recover critical data but for
> ordinary conversational text it is plenty reliable.

Yeah the only problem with that is that your further weakening
an already very weak CRC for use on HF.

I've seen undetected errors already without stunts like that.

Tom

22. Re: [HFSIG:2943] Bad Packets

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 5 Nov 1998 19:02:43 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <m0zbcAd-0014dBC@maggie.ka9q.ampr.org>
Rendered body type: text/plain

Brian Beezley wrote:

  >Yes, there is no way to correct the packets with wrong CRC (or
  >at least - I wouldn□t do it with just 16 bit CRC, too little
  >redundandy).
  >So throw it away...

  I regularly correct bad PACTOR packets by flipping each data bit in turn and
  rechecking for valid 16-bit CRC.

This works, but is dangerous -- it significantly increases the chances of
an undetected error. It would be much better to use a lower-rate code
with sufficient redundancy to provide good error detection *and* correction.
Something like a Reed-Solomon code, say. A typical RS code can correct
up to half as many errored symbols per block as there are parity symbols,
yet maintain an extremely low undetected error rate.

Phil

23. Re: [HFSIG:2942] Re: Kiss

From: Phil Karn <karn@qualcomm.com>
Date: Thu, 5 Nov 1998 19:03:32 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <m0zbcBQ-0014dIC@maggie.ka9q.ampr.org>
Rendered body type: text/plain

Alexander Kurpiers wrote:

  BTW I still doubt that AX25 is a good protocol to use it together with
  HF modems.

Absolutely true. And KISS is a hack that K3MC and I devised 12 years
ago to handle a very specific problem -- getting "raw" frames between
a TNC and a PC that didn't have its own HDLC interface. It really
wasn't meant to be used for anything else.

There are many problems with HDLC over radio channels. The flag
is much too short; random noise fed to an HDLC receiver will regularly
generate spurious flags. The CRC is too small, meaning it's not at
all impossible for a frame of pure noise to pass the CRC check.
There's no facility for error correction. The omission of FEC is
fundamental; it cannot be fixed by layering anything above HDLC.

HDLC may be usable as a higher level framing scheme if you put
enough *under* it to provide robust synchronization, interleaving, FEC
and the like. But once you do all that, there's not much value added
by the HDLC layer anyway. You might as well just define your own
packet format.

Phil

24. Re: [HFSIG:2946] Re: HF digital activity is growing!

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Fri, 06 Nov 1998 08:23:13 +0100
To: hfsig@tapr.org
Message-ID: <3642A3E1.227F3F0D@ife.ee.ethz.ch>
Rendered body type: text/plain

Wheatley, Maurice wrote:

> Most hardware and software engineers use "black boxes" all the time. They
> are called IC's. How many people design with these black boxes and don't

And how many times have you been bitten by not having enough
documentation
on a chip? It happened to me more than once, for example with one of
ADI's DDS chips, in the beginning they had very inadequate
documentation.
Fortunately, they changed it.

Tom

25. Re: [HFSIG:2950] Re: HF digital activity is growing!

From: malberts@ncs.com.na (Mike Alberts)
Date: Fri, 06 Nov 1998 09:35:47 +0200
To: hfsig@tapr.org
Message-ID: <3642A6D3.8BA6715C@ncs.com.na>
Rendered body type: text/plain

Thomas Sailer wrote:

> Wheatley, Maurice wrote:
>
> > Most hardware and software engineers use "black boxes" all the time. They
> > are called IC's. How many people design with these black boxes and don't
>
> And how many times have you been bitten by not having enough
> documentation
> on a chip? It happened to me more than once, for example with one of
> ADI's DDS chips, in the beginning they had very inadequate
> documentation.
> Fortunately, they changed it.
>
> Tom

I agree ,

Documentation is sometimes a problem , and we in Africa are hit the
hardest....NO ARRL or such and support groups to echange ideas....what about
those NOT having internet and e-mail etc..????
It's now the 4th time I get morse code software without documentation...Can
anyone help to get some interface diagram to do Morse code from the PC....????

Thanks & 73's
Mike V51MA

26. Re: [HFSIG:2949] Re: Kiss

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Fri, 06 Nov 1998 09:51:53 +0000
To: hfsig@tapr.org
Message-ID: <3642C6B9.6C02@marconicomms.com>
Rendered body type: text/plain

Phil Karn wrote:

>
> HDLC may be usable as a higher level framing scheme if you put
> enough *under* it to provide robust synchronization, interleaving, FEC
> and the like. But once you do all that, there's not much value added
> by the HDLC layer anyway. You might as well just define your own

> packet format.
>
> Phil

That's the conclusion I came to. Drop the HDLC.
HF modems usually have a sync sequence that gets you
byte sync. If there's interleave or FEC you've got to
sync that anyway. So you know when the 1st packet starts.
HF modems also have mechanisms for determining the end
of the transmission. So you know when the last packet
ends. They need a similar mechanism put in to separate
the packets.

Most modem FEC schemes are intended for use on fixed
block sizes. The HDLC packets can be any size, and the
KISS modem has no knowledge how big they are.

This suggests an extension to KISS so that the stack
can select one of several packet sizes which have been
chosen for efficient FEC block codes. If they're a
fixed size the separation of packets does not require
any flags at all; it can be done by byte counting;
and it's much more robust.

Unfortunately I think you still need a method of
putting a small amount of data in a packet if that's
all there is waiting to be sent; but that should be
easy to deal with on packets which have been accepted
as correct.

OK so AX-25 isn't ideal for HF. But the AX-25 stack
is a standard building-block component available on
several platforms. If we can fix the broken bits on
the modem side of the KISS interface then use it.

Ant
--
Recovering from `changed corporate identity syndrome'
_____

A.N.Martin anthony.n.martin@marconicomms.com www.marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
Marconi Communications UK, Marconi House, New St. Chelmsford, UK
_____

27. Re: [HFSIG:2951] Re: HF digital activity is growing!

From: W6RCecilA <w6rca@ibm.net>
Date: Fri, 06 Nov 1998 07:55:42 -0700
To: hfsig@tapr.org
Message-ID: <36430DEE.4F50@ibm.net>
Rendered body type: text/plain

Mike Alberts wrote:
> It's now the 4th time I get morse code software without documentation...Can
> anyone help to get some interface diagram to do Morse code from the PC....????

Is one of those packages SuperMorse? I think the SuperMorse manual is available
in electronic form including an ASCII schematic interface diagram.
--
73, Cecil, W6RCA http://people.delphi.com/CecilMoore

28. Re: [HFSIG:2948] Re: Bad Packets

From: Robert McGwier <rwmcgwier@home.com>
Date: Sun, 08 Nov 1998 09:38:52 -0500
To: hfsig@tapr.org
Message-ID: <3645ACFC.617302F7@home.com>
Rendered body type: text/plain

This does increase the likelihood of passing a bad packet with the
correct CRC. However, if you do the "guessing which bit to correct"
based upon soft decisions and using likelihoods, AND you choose a
CRC that has a large distance given the size packets you are considering
(not much difference in small packets), then you can easily have
a system that allows you to correct one or two errors with greatly
decreased chance of making the type error Phil alludes to.

Phil Karn wrote:

>
> Brian Beezley wrote:
>
> >Yes, there is no way to correct the packets with wrong CRC (or
> >at least - I wouldn´t do it with just 16 bit CRC, too little
> >redundandy).
> >So throw it away...
>
> I regularly correct bad PACTOR packets by flipping each data bit in turn and
> rechecking for valid 16-bit CRC.
>
> This works, but is dangerous -- it significantly increases the chances of
> an undetected error. It would be much better to use a lower-rate code
> with sufficient redundancy to provide good error detection *and* correction.
> Something like a Reed-Solomon code, say. A typical RS code can correct
> up to half as many errored symbols per block as there are parity symbols,
> yet maintain an extremely low undetected error rate.
>
> Phil

29. Re: [HFSIG:2954] Re: Bad Packets

From: Phil Karn <karn@ka9q.ampr.org>
Date: Tue, 10 Nov 1998 13:11:55 -0800
To: hfsig@tapr.org
Message-ID: <3648AC1B.BE935DAC@ka9q.ampr.org>
Rendered body type: text/plain

Robert McGwier wrote:

> This does increase the likelihood of passing a bad packet with the
> correct CRC. However, if you do the "guessing which bit to correct"
> based upon soft decisions and using likelihoods, AND you choose a
> CRC that has a large distance given the size packets you are considering
> (not much difference in small packets), then you can easily have
> a system that allows you to correct one or two errors with greatly
> decreased chance of making the type error Phil alludes to.

Correct. I should note what "large distance" means here -- it means
a large CRC field relative to the packet size. Since the CRC is fixed
and fairly small (16 bits), this implies a small frame size, which
has other problems.

Code distance can be "spent" on error detection, error correction, or both.
Any error correcting scheme necessarily increases the chances
of an undetected error, as opposed to a purely error *detecting* scheme.
If the code has sufficient distance (which translates roughly to sufficient
overhead), the chances of an undetected error can be kept to negligible
levels. And for a constant overhead ratio, bigger blocksizes are better. So
a big Reed-Solomon code can give excellent error correcting ability with an
negligible undetected error rate, but a small block, low redundancy code
like a CRC has very little distance even for error detection, much less
correction.

Phil

30. STANAG-5066 Annex G

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Wed, 11 Nov 1998 19:21:43 +0000
To: hfsig@tapr.org
Message-ID: <3649E3C7.7FAB@marconicomms.com>
Rendered body type: text/plain

STANAG-5066 Annex G (not yet ratified) can be found at
http://www.primenet.com/~moorer/5066ag.zip

It describes serial tone modem waveforms for
coded rates of 3200, 4800, 6400, 9600 bps.

Enjoy :-)

Ant
--
Recovering from `changed corporate identity syndrome'
____

A.N.Martin   anthony.n.martin@marconicomms.com   www.marconicomms.com
Snr. Development Engineer   Tel.   01245 353221 x4995
Military Products Engineering   Fax.   01245 275993
Marconi Communications UK, Marconi House, New St. Chelmsford, UK

_____

31. Re: [HFSIG:2956] STANAG-5066 Annex G

From: forrerj@peak.org (Johan Forrer)
Date: Thu, 12 Nov 1998 21:02:41 -0800
To: hfsig@tapr.org
Message-ID: <199811130506.VAA05395@peak.org>
Rendered body type: text/plain

Anthony,

Thanks for the info. Do you perhaps know where the other lower-speed
STANAG 4285 info is located on the web? I believe there is a document
that covers the characteristics of the 1200/2400/3600 bps operation.

Regarding 4285 --- If I understand things correctly, it operates at 2400
baud symbol rate, which I am quite sceptical about. I rather like
STANAG 4529 for this purpose -- 75,150, 300, 600, 1200
coded, much more robust and especially since it operates in
1240 Hz wide channels.

Regards,

Johan, KC7WW

>STANAG-5066 Annex G (not yet ratified) can be found at
>http://www.primenet.com/~moorer/5066ag.zip
>
>It describes serial tone modem waveforms for
>coded rates of 3200, 4800, 6400, 9600 bps.
>
>Enjoy :-)
>
>Ant
>--
>Recovering from `changed corporate identity syndrome'
>_____
>
>A.N.Martin   anthony.n.martin@marconicomms.com   www.marconicomms.com
>Snr. Development Engineer   Tel.   01245 353221 x4995
>Military Products Engineering   Fax.   01245 275993
>Marconi Communications UK, Marconi House, New St. Chelmsford, UK
>_____
>
>
>

32. Re: [HFSIG:2957] Re: STANAG-5066 Annex G

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Fri, 13 Nov 1998 09:54:01 +0000
To: hfsig@tapr.org
Message-ID: <364C01B9.9DD@marconicomms.com>
Rendered body type: text/plain

Johan Forrer wrote:
>
> Anthony,
>
> Thanks for the info. Do you perhaps know where the other lower-speed
> STANAG 4285 info is located on the web? I believe there is a document
> that covers the characteristics of the 1200/2400/3600 bps operation.
>
Yes it basically covers 1200 2400 & 3600 bps uncoded. The on-air
symbol rate is 2400 bps and with 8-psk modulation. Half the on-air
data rate is taken up with fixed bits.

Annex E describes the convolutional interleaver and convolutional
encoding for 75, 150, 300, 600, 1200 & 2400 bps.

I haven't seen it or heard any URLs for '4285 spec on my travels.

I have access to paper copy of 4285. If I had an electronic copy
I'd email it to you.

> Regarding 4285 --- If I understand things correctly, it operates at 2400
> baud symbol rate, which I am quite sceptical about. I rather like
> STANAG 4529 for this purpose -- 75,150, 300, 600, 1200
> coded, much more robust and especially since it operates in

> 1240 Hz wide channels.

STANG-4529 is essentially a half-rate '4285. But they reduced the BW
a little further still which doesn't do it any good at all.

The problem with the design of 4529 is that the doppler capability is
reduced by slightly more than a factor of 2 relative to 4285. That
does not make it robust. The interleavers end up twice as long and
a 20 second interleaver does help if you can wait that long. The
1200 bps mode is reckoned to have `limited utility for skywave
operation'. I think it's a sow's ear. My copy is version 1.1 - 8
May 1995; it could have been changed since and my comments may not
apply.

4285 is more robust than 4529.

If you want robust, STANAG-4415 is your man. It can cope with more
multipath delay and more doppler spread than the worst high-latitude
links encountered by the DAMSON sounding project. The only thing that
stops it working is too low SNR, ie no signal.

--
Recovering from `changed corporate identity syndrome'
____

A.N.Martin anthony.n.martin@marconicomms.com www.marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
Marconi Communications UK, Marconi House, New St. Chelmsford, UK
____

33. STANAG 4285

From: "Furman, William" <WFurman@harris.com>
Date: Fri, 13 Nov 1998 07:35:39 -0500
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <51650CE89911D111B4540000F840E39802172712@rfcmx2.rfc.comm.harris.com>
Rendered body type: text/plain

Hello All,

STANAG 4285 contains an annex, annex E which specifies FEC Coding
interleaving, providing user bit rates of 75-> 2400 bps.

Both 4285 and 188-110A utilize M-PSK signalling at 2400 symbols per
second.
The PSK is spectrally constrained by filtering prior to quadrature
modulating an
1800 Hz sub-carrier.

Regards,

Bill

==
William N. Furman (716)-242-3553
Sr. Principal Engineer (716)-242-3217 (Fax)
Harris Corporation wfurman@harris.com
RF Communications Division
1680 University Avenue
Rochester, New York, 14610

34. STANAG 4285

From: "Marcelo Puhl" <mark@plug-in.com.br>
Date: Sat, 14 Nov 1998 15:10:39 -2
To: hfsig@tapr.org
Message-ID: <199811141815.PAA16164@bigbox.plug-in.com.br>
Rendered body type: text/plain

>
> Hello All,
>
> STANAG 4285 contains an annex, annex E which specifies FEC Coding
> interleaving, providing user bit rates of 75-> 2400 bps.
>
> Both 4285 and 188-110A utilize M-PSK signalling at 2400 symbols per
> second.
> The PSK is spectrally constrained by filtering prior to quadrature
> modulating an
> 1800 Hz sub-carrier.

>
> Regards,
>
> Bill
>
> ======================================
> William N. Furman   (716)-242-3553
> Sr. Principal Engineer   (716)-242-3217 (Fax)
> Harris Corporation   wfurman@harris.com
> RF Communications Division
> 1680 University Avenue
> Rochester, New York, 14610
>

Hi,

How implement that on a SoundBlaster ? It would be possible ?

73,
    Marcelo - PY3SS

35. Re: [HFSIG:2960] STANAG 4285

From: forrerj@peak.org (Johan Forrer)
Date: Sat, 14 Nov 1998 10:51:17 -0800
To: hfsig@tapr.org
Message-ID: <199811141854.KAA03523@peak.org>
Rendered body type: text/plain

Marcelo,

That would nice, probably quite possible.

To get an idea on what is involved doing such an implementation on a Pentium
class
processor and soundblaster, have a look at Hansi Reiser's implementation of
PSK31
that is available on Eduardo's web page.
(http://det.bi.ehu.es/~jtpjatae/ham.html).

PSK31 is not as complex as STANAG 4285, but it implements both BPSK, and
QPSK and includes Viterbi decoding (softdecisions) . There are implementations
available for TI320C50, M56002, SHARC, and for Linux using a soundcard.
PSK31 has quite a following in Europe, and I can vouch for it having excellent
performance from on-the-air tests that WB6RIJ and I have done over the past
year or so.

Being Linux, Hansi's implementation provides all the source code for you to
see how
hes done it -- in this case it is done all in C++, although the
implementation is not a pure
object-oriented design, it is merely to make code more readable. It is not
too difficult
to follow and it works real nicely on the air too!

This might be a starting point for your 4285 implementation?

73

Johan

>>
>> Hello All,
>>
>> STANAG 4285 contains an annex, annex E which specifies FEC Coding
>> interleaving, providing user bit rates of 75-> 2400 bps.
>>
>> Both 4285 and 188-110A utilize M-PSK signalling at 2400 symbols per
>> second.
>> The PSK is spectrally constrained by filtering prior to quadrature
>> modulating an
>> 1800 Hz sub-carrier.
>>
>> Regards,
>>
>> Bill
>>
>> ======================================

>> William N. Furman   (716)-242-3553
>> Sr. Principal Engineer   (716)-242-3217 (Fax)
>> Harris Corporation   wfurman@harris.com
>> RF Communications Division
>> 1680 University Avenue
>> Rochester, New York, 14610
>>
>
>Hi,
>
>How implement that on a SoundBlaster ? It would be possible ?
>
>---
>73,
>   Marcelo - PY3SS
>
>
>

36. Re: [HFSIG:2961] Re: STANAG 4285

From: Charles Brain <chbrain@dircon.co.uk>
Date: Sun, 15 Nov 1998 08:54:42 +0000
To: hfsig@tapr.org
Message-ID: <364E96D2.FB76485D@dircon.co.uk>
Rendered body type: text/plain

Johan Forrer wrote:
>
> Marcelo,
>
> That would nice, probably quite possible.
>

> >
> >Hi,
> >
> >How implement that on a SoundBlaster ? It would be possible ?
> >
> >---
> >73,
> >   Marcelo - PY3SS

Hello all,

Below is an excerpt from an email I received earlier in the year.
HF98 is a Scandinavian Professional HF radio conference.

--------------------------XXXXXXXXXXXXXXX----------------------------
There are some interesting HF 98 papers on really high speed modems
(16 kbps in 6 kHz BW up to 64 kbps @ 8 kHz BW) which I think is the
way forward. I guess you know that Harris has implemented the 110A
single tone modem on a Pentium PC.
--------------------------XXXXXXXXXXXXXX----------------------------

So it looks as if Harris (HINT!) have done this for the MIL-STD.
I doubt it is DOS or Windows based!

Johan,

If you still want it I think I have an electronic copy of STANAG 4285,
I think I have it on one of my back-up CDROMs, I will have a look
for it, it is quite an old copy however. I think it came from
Peter Reynolds page at SHAPE Technical centre .
I have lost the URL sri.

- Charles G4GUO

37. Re: [HFSIG:2962] Re: STANAG 4285

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Mon, 16 Nov 1998 12:21:32 +0000
To: hfsig@tapr.org
Message-ID: <365018CC.7C32@marconicomms.com>
Rendered body type: text/plain

Charles Brain wrote:
>
> --------------------------XXXXXXXXXXXXXXX----------------------------
> There are some interesting HF 98 papers on really high speed modems
> (16 kbps in 6 kHz BW up to 64 kbps @ 8 kHz BW) which I think is the

> way forward. I guess you know that Harris has implemented the 110A
> single tone modem on a Pentium PC.
> -------------------------XXXXXXXXXXXXX----------------------------
>
I didn't go to HF98 but I've borrowed the proceedings from
someone who did. It's full of good stuff as usual.

There's a paper on a 16kbps modem by N.C. Davies of DERA
and R.W. Johnson & M.B. Jorgenson Comms Research Centre, Canada.

It is designed for a 6kHz channel BW and can be used with
6kHz SSB or 2 x 3kHz ISB equipment. For ISB it modulates
two carriers at half the rate.

The makeup of the on-air waveform has mini-probes and re-inserted
preambles similar to the usual serial waveforms. It uses yet
another variant of the 16-QAM constellation. 0.1s and 6.5s
interleavers are used.

The results show it supports 2Hz doppler spread and 2ms
multipath (SSB variant) and 2Hz / 4ms (ISB variant).
Whether you think that's adequate is up to you.

It is implemented on a Pentium using the QNX realtime OS
and `an inexpensive soundcard'.

There's also a paper on 125kHz BW OFDM with MAP decoding
for `stealth' (low probability of intercept) applications.

There's another paper from Harris on a waveform for
2.4kbps to 64kbps using trellis-coded OFDM using BWs
from 3kHz to 80kHz. Has been tried on 170km NVIS path
and 175km surface-wave over seawater using frequencies
close to 30MHz. It was implemented using 8 x Texas 'C44 DSP.

MIL-STD-188-110A and STANAG-4285 et al can
all be implemented with a Pentium CPU. The choice of
equaliser has a major influence on the cycles required.
If considering doing it within Win 95/98/NT, then
then there are many issues surrounding efficiency of the
OS, ring-fencing the required cycles for the modem,
and whether the machine remains usable for applications,
network access etc.

If you really want to do modems on a pentium & inside
your PC, look in any industrial PC catalogue and you
will find plug-in PCI-bus pentium SBC cards. Consider
adding one of those. No doubt they offer more MFLOPS
for your money than than similar class DSP cards.

If you find the electronic copy of STANAG-4285 Charles,
pls e-mail me a copy too!

--
Recovering from `changed corporate identity syndrome'
____

A.N.Martin anthony.n.martin@marconicomms.com www.marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
Marconi Communications UK, Marconi House, New St. Chelmsford, UK
____

38. [HFSIG:2963] Re: STANAG 4285 -Reply

From: Danie Brynard <BRYD@kidd.Co.Za>
Date: Tue, 17 Nov 1998 11:32:55 +0200
To: hfsig@tapr.org
Message-ID: <s6515ef0.028@kidd.Co.Za>
Rendered body type: text/plain

You mention the industrial type SB cards. What DSP do they have in ?
Can one do development for it ?

Danie

>>> Anthony Martin <anthony.n.martin@MARCONICOMMS.COM>

16/November/1998 02:24pm >>>
Charles Brain wrote:
>
> -------------------------XXXXXXXXXXXXXXX-------------------------
> There are some interesting HF 98 papers on really high speed modems
> (16 kbps in 6 kHz BW up to 64 kbps @ 8 kHz BW) which I think is the
> way forward. I guess you know that Harris has implemented the 110A
> single tone modem on a Pentium PC.
> -------------------------XXXXXXXXXXXXXX-------------------------
>
I didn't go to HF98 but I've borrowed the proceedings from
someone who did. It's full of good stuff as usual.

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and R.W. Johnson & M.B. Jorgenson Comms Research Centre, Canada.

It is designed for a 6kHz channel BW and can be used with
6kHz SSB or 2 x 3kHz ISB equipment. For ISB it modulates
two carriers at half the rate.

The makeup of the on-air waveform has mini-probes and re-inserted
preambles similar to the usual serial waveforms. It uses yet
another variant of the 16-QAM constellation. 0.1s and 6.5s
interleavers are used.

The results show it supports 2Hz doppler spread and 2ms
multipath (SSB variant) and 2Hz / 4ms (ISB variant).
Whether you think that's adequate is up to you.

It is implemented on a Pentium using the QNX realtime OS
and `an inexpensive soundcard'.

There's also a paper on 125kHz BW OFDM with MAP decoding
for `stealth' (low probability of intercept) applications.

There's another paper from Harris on a waveform for
2.4kbps to 64kbps using trellis-coded OFDM using BWs
from 3kHz to 80kHz. Has been tried on 170km NVIS path
and 175km surface-wave over seawater using frequencies
close to 30MHz. It was implemented using 8 x Texas 'C44 DSP.

MIL-STD-188-110A and STANAG-4285 et al can
all be implemented with a Pentium CPU. The choice of
equaliser has a major influence on the cycles required.
If considering doing it within Win 95/98/NT, then
then there are many issues surrounding efficiency of the
OS, ring-fencing the required cycles for the modem,
and whether the machine remains usable for applications,
network access etc.

If you really want to do modems on a pentium & inside
your PC, look in any industrial PC catalogue and you
will find plug-in PCI-bus pentium SBC cards. Consider
adding one of those. No doubt they offer more MFLOPS
for your money than than similar class DSP cards.

If you find the electronic copy of STANAG-4285 Charles,
pls e-mail me a copy too!

--
Recovering from `changed corporate identity syndrome'
_____

A.N.Martin anthony.n.martin@marconicomms.com
www.marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
Marconi Communications UK, Marconi House, New St. Chelmsford, UK
_____

39. Re: [HFSIG:2964] Re: STANAG 4285 -Reply

From: Anthony Martin <anthony.n.martin@marconicomms.com>
Date: Tue, 17 Nov 1998 15:53:46 +0000
To: hfsig@tapr.org
Message-ID: <36519C0A.54B2@marconicomms.com>

Rendered body type: text/plain

Danie Brynard wrote:
>
> You mention the industrial type SB cards. What DSP do they have in ?
> Can one do development for it ?
>

>>
>> If you really want to do modems on a pentium & inside
>> your PC, look in any industrial PC catalogue and you
>> will find plug-in PCI-bus pentium SBC cards. Consider
>> adding one of those. No doubt they offer more MFLOPS
>> for your money than than similar class DSP cards.

I said SBC = Single Board Computer.

(not Sound Blaster)

We have far too many acronyms these days!

Many industrial PCs are based on a passive backplane with
15 or more slots. The CPU is not on the motherboard but on
a plug-in card. These systems can have stacks of IO or
several independent processor cards in the same box.

My suggestion is that you could plug an extra pentium
processor card into a PC with separate RAM etc to run modem
software without worrying about the OS or applications on
the main processor. And such pentium-based processor cards
would cost a lot less than a comparable card using SHARCs or
TMS320C6xxx as they're made in higher volumes and they're
based on standard PC chipsets.

Dual-pentium motherboards are only a little more expensive
than single-processor ones. But that doesn't enable you to
run Windows on one processor and something else suitable
for real-time on the other.

--
Recovering from `changed corporate identity syndrome'
____

A.N.Martin anthony.n.martin@marconicomms.com www.marconicomms.com
Snr. Development Engineer Tel. 01245 353221 x4995
Military Products Engineering Fax. 01245 275993
Marconi Communications UK, Marconi House, New St. Chelmsford, UK
____

40. VHF Receiver

From: malberts@ncs.com.na (Mike Alberts)
Date: Wed, 25 Nov 1998 15:34:44 +0200
To: HF mail <hfsig@tapr.org>
Message-ID: <365C0774.ECDA0F5F@ncs.com.na>
Rendered body type: text/plain

Hi
URGENT.

Can anyone help with a simple AM receiver circuit....Frequency should be
123.1 MHZ.
Minimum components required and output must be to a lightweight
headset...
Thanks
Mike V51MA

41. Re: VHF Receiver

From: "Fernando M. I. Carreiro" <fmi@carreiro.com>
Date: Wed, 25 Nov 1998 23:34:36 -0000
To: <hfsig@tapr.org>
Message-ID: <003601be18cc$9f0d5a40$e6ffa8c0@pc001>
Rendered body type: text/plain

On Wednesday, 25 November, 1998 13:54, Mike Alberts wrote:
>Can anyone help with a simple AM receiver circuit....

    Yes, here is some info. If small parts count is required, low price and small footprint, you might consider building a circuit around the RF2906 from RF Micro Devices. You will find full datasheet specs at their web site "http://www.rfmd.com". This little device is a full transeiver for 30MHz to 600Mhz.

Will do AM/FM as well as OOK, ASK, FSK and BPSK. It can be driven from 2.7V to 5.5V battery supply at 4.6mA(3V) for receive mode (-97dBm Sensitivity). All you need is the passive components for tuning and and audio amplifier to drive the head-phones.

     You should take a look at their Web Site as they have other devices which may be of interest. I currently using this device for low power data transmission between PalmPilots on European ISM bands. The RF2906 cost me roughly $10 to $15 per unit (depending on exchange rate).

Fernando M. I. Carreiro
mailto:fmi@carreiro.com
http://www.fmi.carreiro.com
callsign:ZS6BLF/CT1HET

-----Original Message-----
From: Mike Alberts <malberts@ncs.com.na>
To: hfsig@tapr.org <hfsig@tapr.org>
Date: Wednesday, 25 November, 1998 13:54
Subject: [HFSIG:2966] VHF Receiver

>Hi
>URGENT.
>
>Can anyone help with a simple AM receiver circuit....Frequency should be
>123.1 MHZ.
>Minimum components required and output must be to a lightweight
>headset...
>Thanks
>Mike   V51MA
>
>

42. New version of Hamview

From: Alberto di Bene <dibene@usa.net>
Date: 27 Nov 98 10:32:50 MET
To: hfsig@tapr.org
Cc: dsp@tapr.org
Message-ID: <19981127093250.25609.qmail@www0i.netaddress.usa.net>
Rendered body type: text/plain

Version 2.10 of Hamview has been just uploaded to :
   http://www.freeyellow.com/members/padan

   HAMVIEW is a freeware program that:

- gets an audio signal either from a Sound Blaster(TM) compatible card or from a .WAV file
- performs a windowed spectral analysis on the incoming signal
- displays the signal spectrum on the video screen using a carefully matched color scheme
- performs some enhancement on the signal using signal processing methods
- plays the filtered signal at the output of the audio card
- and last but not least, all these goodies are done in real time! (well, to be honest with about one second of delay from the input to the output, due to the data windows used for signal analysis and processing).

   The program was especially developed for radio amateurs interested into all those communication techniques where signals very close to (or more often below) the receiver noise floor are used, like EME, VLF, and so on.

   All the hardware needed is:

- a Pentium computer with MS-DOS or PC-DOS 6.xx or greater (a W95 DOS session is OK). Windows NT is not supported, as the program needs to talk directly to the hardware, bypassing any driver, and this is not allowed under NT.
- a Creative SB16 sound card or better. Compatible cards should work, provided that 100% hardware compatibility is assured, and the card is full-duplex compatible.

   What's new in version 2.10 ?

- New INI file management (many more settings are now remembered)
- Passthrough mode (useful when tuning, as the audio delay is suppressed)
- Now both output channels are driven (this was a bug in the previous versions)
- The output volume is adjustable with cursor Up and Down keys
- The local time is diplayed at the full minute
- New graphics library, which means more compatibility with various video

cards
- The East/West indicator has been moved to the bottom of the display
- Improved .WAV files management
- Medium speed and Fast speed modes added, dynamically changeable from the
keyboard
- A few bugs have been exterminated

   Enjoy, and, please, do bugs reporting ! Thanks.

   73,
   Alberto di Bene
   I2PHD

_____
Get free e-mail and a permanent address at http://www.netaddress.com/?N=1

43. Re: [HFSIG:2968] New version of Hamview

From: "Jerome M. Kutche" <n9lya@tima.com>
Date: Fri, 27 Nov 1998 17:39:55 -0500
To: hfsig@tapr.org, dibene@usa.net
Message-ID: <365F2A37.40BB3DB9@tima.com>
Rendered body type: text/plain

Hi I downloaded Hamview, Read the readme and it is not working... I have all the
min req.
I typed
hamview.exe -v0 -m
I am using 640x480 256
I am sure it is something I am doing wrong...
Can you advise

Many Thanks... Jerry

Alberto di Bene wrote:

> Version 2.10 of Hamview has been just uploaded to :
>   http://www.freeyellow.com/members/padan
>
>   HAMVIEW is a freeware program that:
>
> - gets an audio signal either from a Sound Blaster(TM) compatible card or from
> a .WAV file
> - performs a windowed spectral analysis on the incoming signal
> - displays the signal spectrum on the video screen using a carefully matched
> color scheme
> - performs some enhancement on the signal using signal processing methods
> - plays the filtered signal at the output of the audio card
> - and last but not least, all these goodies are done in real time! (well, to
> be honest with about one second of delay from the input to the output, due to
> the data windows used for signal analysis and processing).
>
>   The program was especially developed for radio amateurs interested into all
> those communication techniques where signals very close to (or more often
> below) the receiver noise floor are used, like EME, VLF, and so on.
>
>   All the hardware needed is:
>
> - a Pentium computer with MS-DOS or PC-DOS 6.xx or greater (a W95 DOS session
> is OK). Windows NT is not supported, as the program needs to talk directly to
> the hardware, bypassing any driver, and this is not allowed under NT.
> - a Creative SB16 sound card or better. Compatible cards should work, provided
> that 100% hardware compatibility is assured, and the card is full-duplex
> compatible.
>
>   What's new in version 2.10 ?
>
> - New INI file management (many more settings are now remembered)
> - Passthrough mode (useful when tuning, as the audio delay is suppressed)
> - Now both output channels are driven (this was a bug in the previous
> versions)
> - The output volume is adjustable with cursor Up and Down keys
> - The local time is diplayed at the full minute
> - New graphics library, which means more compatibility with various video
> cards
> - The East/West indicator has been moved to the bottom of the display

> - Improved .WAV files management
> - Medium speed and Fast speed modes added, dynamically changeable from the
> keyboard
> - A few bugs have been exterminated
>
>   Enjoy, and, please, do bugs reporting ! Thanks.
>
>   73,
>   Alberto di Bene
>   I2PHD
>
> ____
> Get free e-mail and a permanent address at http://www.netaddress.com/?N=1

--
____
=NOTICE TO BULK E-MAILERS PLEASE READ THIS BEFORE BULK E-MAILING THIS
ADDRESS=
Pursuant to US Code, Title 47, Chapter 5, Subchapter II, p.227, any and all
non-solicited commercial E-mail sent to this address is subject to a
download and archival fee in the amount of $500 US. Anyone who sends
unsolicited commercial e-mail to this account will be charged a $500
proofreading fee. Consider this official notification. Failure to abide
by this will result in legal action.
For a complete summary of this Legislation see

http://thomas.loc.gov/cgi-bin/bdquery/z?d105:SN01618:@@@D
____

44. Re: [HFSIG:2968] HFSIG digest 602

From: "S.A.Monteiro" <sermon@cpovo.net>
Date: Fri, 27 Nov 1998 21:41:38 -0200
To: hfsig@tapr.org
Message-ID: <365F2AA2.530780D0@cpovo.net>
Rendered body type: text/plain

hfsig@tapr.org escreveu:

>   HFSIG Digest 602
>
> Topics covered in this issue include:
>
>   1) New version of Hamview
>   by Alberto di Bene <dibene@usa.net>
>
> --

Olá. Peguei o programa, mas infelizmente, como aconteceu com outros, a minha
SBlaster se recusa a funcionar com programas em DOS.
O único que trabalha é o BTL, qualquer outro não sendo do W95 não tem jeito. E não
adianta setar aquele cmd blaster= axxx iy dz no autoexec.
É uma pena, pq esse hamview parecia interessante. Bye

45. Re: [[HFSIG:2970] Re: HFSIG digest 602]

From: Alberto di Bene <dibene@usa.net>
Date: 27 Nov 98 23:26:02 MET
To: hfsig@tapr.org
Message-ID: <19981127222602.3456.qmail@www0s.netaddress.usa.net>
Rendered body type: text/plain

"S.A.Monteiro" <sermon@cpovo.net> wrote:

> Ol□. Peguei o programa, mas infelizmente, como aconteceu com outros, a >
minha
> SBlaster se recusa a funcionar com programas em DOS.
> O □nico que trabalha □ o BTL, qualquer outro n□o sendo do W95 n□o tem >
jeito. E n□o adianta setar aquele cmd blaster= axxx iy dz no autoexec.
> □ uma pena, pq esse hamview parecia interessante. Bye
>

 Sorry, but all what I was able to understand is that you have problems with
the BLASTER settings. I don't read Portuguese. If you can send me a message in
English (or Italian, or French) maybe I can help you.

73,
Alberto di Bene
I2PHD

____

Get free e-mail and a permanent address at http://www.netaddress.com/?N=1

46. Re: [[HFSIG:2969] Re: New version of Hamview]

From: Alberto di Bene <dibene@usa.net>
Date: 28 Nov 98 01:30:03 MET
To: hfsig@tapr.org
Message-ID: <19981128003003.18753.qmail@www08.netaddress.usa.net>
Rendered body type: text/plain

> Hi I downloaded Hamview, Read the readme and it is not working... I have all
the
> min req.
> I typed
> hamview.exe -v0 -m
> I am using 640x480 256
> I am sure it is something I am doing wrong...

> Can you advise
>
> Many Thanks... Jerry

Replied offline. Please do not send bugs reports or queries for help to the
list, but instead to me directly, at dibene@usa.net
Thanks,

Alberto,
I2PHD

_____
Get free e-mail and a permanent address at http://www.netaddress.com/?N=1

47. 1998 ARRL and TAPR DCC Audio, Photos, Abstracts, and More

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Mon, 30 Nov 1998 15:25:13 -0600
To: "HF SIG list mailing"<hfsig@tapr.org>, " Spread Spectrum "<ss@tapr.org>,
" TAPR/AMSAT DSP "<dsp@tapr.org>,
"APRS SIG list mailing"<aprssig@tapr.org>,
"NETSIG list mailing"<netsig@tapr.org>,
"BBS SIG list mailing"<bbssig@tapr.org>,
" tacgps "<tacgps@tapr.org>, aprsnews, mic-e,
TAPR Regional Freq <regional_freq@tapr.org>,
"TAPR-BB list mailing"<tapr-bb@tapr.org>
Message-ID: <v04020a06b288bb17413f@[128.83.74.104]>
Rendered body type: text/plain

The ARRL and TAPR DCC Web page has been updated to include all recorded
segments from the conference as well as a story and photo section.

Visit: http://www.tapr.org/dcc/

* Listen to sessions from the 1998 ARRL and TAPR DCC over 30 hours covering
Keynote by Dale Hatfield, W0IFO
Chief of the FCC's Office of Engineering and Technology
APRS National Symposium
Packet Radio Users Group of Japan (PRUG) Technical Seminar
Technical Paper Presentations
Introductory Sessions
Introduction to Spread Spectrum
Introduction to PIC Development
Introduction to MMDS/LMDS
Friday Technical Seminar featuring Don Lemke, WB9MJN on
Infrastructureless Packet Radio Networks
Sunday Technical Seminars featuring:
Lyle Johnson, WA7GXD -
RUDAK digital communications system
Tim Shepard, KD1YK -
Packet Radio Networks with Millions or Billions of
Stations

* 1998 Conference Overview Story

* Look at photos from the 1998 ARRL and TAPR DCC

* Review Abstracts from 1998 Proceedings

* Check out past Digital Communication Conferences Audio and Abstracts

as of now, the 1999 ARRL and TAPR DCC will be hosted in Phoenix, AZ in Sept
'99. More information will be posted once a date and location has been
decided.

Tucson Amateur Packet Radio
8987-309 E Tanque Verde Rd #337 * Tucson, Az * 85749-9399
---
e-mail: tapr@tapr.org   web: <http://www.tapr.org/>
ftp: ftp.tapr.org
phone: 940-383-0000   fax: 940-565-2544

December 1998

Original source file: hfsig.9812
Source: http://ftp.tapr.org/tapr_history/SIG_Archives/hfsig/mail_archive/hfsig.9812

1. Re: [HFSIG:2973] HFSIG digest 604

From: Steve Crouch <Crouchs2@compuserve.com>
Date: Wed, 2 Dec 1998 15:39:55 -0500
To: hfsig <hfsig@tapr.org>
Message-ID: <199812021540_MC2-623F-9640@compuserve.com>
Rendered body type: text/plain

REMOVE

2. Satelite software

From: malberts@ncs.com.na (Mike Alberts)
Date: Fri, 04 Dec 1998 13:03:51 +0200
To: HF mail <hfsig@tapr.org>
Message-ID: <3667C197.1002AEB5@ncs.com.na>
Rendered body type: text/plain

Hi, Could anyone advise where to find satelite software for tracking and
monitoring , and in particular the Spacfe shuttle missions.
Thank you
Mike
V51MA

3. Re: [HFSIG:2975] Satelite software

From: Rob Glassey <robglassey@clear.net.nz>
Date: Sat, 05 Dec 1998 09:53:02 -0800
To: hfsig@tapr.org
Message-ID: <366972FE.6F1E@clear.net.nz>
Rendered body type: text/plain

Mike, V51MA, wrote:

> Hi, Could anyone advise where to find satelite software for tracking
> and monitoring , and in particular the Spacfe shuttle missions.

Try WinOrbit:

ftp://ftp.funet.fi/pub/ham/satellite/tracking/winorb35.zip

463 Kbytes

This is windows based and works very well.

Cheers,

Rob

4. Re: [HFSIG:2976] Re: Satelite software

From: "Leo (Bing) Whiteway" <ve7uw@rac.ca>
Date: Fri, 04 Dec 1998 22:24:18 -0500
To: Gerry Manus <jmanus@telusplanet.net>, hfsig@tapr.org
Message-ID: <3668A762.6D22@rac.ca>
Rendered body type: text/plain

Rob Glassey wrote:
>
> Mike, V51MA, wrote:
>
> > Hi, Could anyone advise where to find satelite software for tracking
> > and monitoring , and in particular the Spacfe shuttle missions.
>
> Try WinOrbit:
>
> ftp://ftp.funet.fi/pub/ham/satellite/tracking/winorb35.zip
>
> 463 Kbytes
>
> This is windows based and works very well.
>
> Cheers,
>
> Rob

--

5. Re: [HFSIG:2976] Re: Satelite software

From: malberts@ncs.com.na (Mike Alberts)
Date: Sat, 05 Dec 1998 09:41:57 +0200
To: hfsig@tapr.org
Message-ID: <3668E3C5.CCAA14F0@ncs.com.na>
Rendered body type: text/plain

Rob Glassey wrote:

> Mike, V51MA, wrote:
>
> > Hi, Could anyone advise where to find satelite software for tracking
> > and monitoring , and in particular the Spacfe shuttle missions.
>
> Try WinOrbit:
>
> ftp://ftp.funet.fi/pub/ham/satellite/tracking/winorb35.zip
>
> 463 Kbytes
>
> This is windows based and works very well.
>
> Cheers,
>
> Rob

Hi Rob,
I have errors downloading this file....give me a website address to go
to....

Thank you
Mike

6. R: [HFSIG:2975] Satelite software

From: "Nicola Savella" <savella@ieee.org>
Date: Sat, 5 Dec 1998 10:55:14 +0100
To: <hfsig@tapr.org>
Message-ID: <01be2035$5b0fa4e0$0100007f@vectra200-vl5>
Rendered body type: text/plain

Hi Mike,
you can find a satelite software searching for STSPLUS at:
http://www.dransom.com/
or directly to:
http://users.sedona.net/~dransom/stsplus.html
the program is STSPLUS and it works in DOS-mode. You can have it downloading two *.zip files.
It is easy to use and you have only to configurate some personal information (as your geographical
position).
The information requested from the program STSPLUS about one particular satelite is a simple two line
elements set
recorded in a file *.txt that you can find at
http://www.celestrak.com/
and go to -Current Data-
They are updated frequently.

73 de Nicola
Patent 3321/TS/W (waiting for Licence)

==========
Nicola Savella
http://space.tin.it/io/nsavell
Biomedical Electronic Engineering - DEEI
University of Trieste - ITALY
Email: savella@ieee.org
==========

>-----Original Message-----
>From: Mike Alberts <malberts@ncs.com.na>
>To: hfsig@tapr.org <hfsig@tapr.org>
>Date: Friday, December 04, 1998 12:18 PM
>Subject: [HFSIG:2975] Satelite software
>
>Hi, Could anyone advise where to find satelite software for tracking and
>monitoring , and in particular the Spacfe shuttle missions.
>Thank you
>Mike
>V51MA
>

7. Re: [HFSIG:2978] Re: Satelite software

From: Rob Glassey <robglassey@clear.net.nz>
Date: Mon, 07 Dec 1998 00:40:03 -0800
To: hfsig@tapr.org
Message-ID: <366B9463.5C43@clear.net.nz>
Rendered body type: text/plain

Mike Alberts wrote:
> > Try WinOrbit:
> >
> > ftp://ftp.funet.fi/pub/ham/satellite/tracking/winorb35.zip
> >
> > 463 Kbytes
> Hi Rob,
> I have errors downloading this file....give me a website address to go
> to....

WinOrbit is by C.D. Gregory, K8CG.

------------------------- From the help -------------------------
Introduction

WinOrbit is a program for computing artificial Earth-satellite position
and visibility, with the Amateur Radio Satellite operator in mind.

WinOrbit is free. If you enjoy the program and want to express your
support, you could join AMSAT, or make a donation to the Amateur
Satellite program, but that's up to you.

Please note: absolutely no guarantees of accuracy or precision are
made!
Commercial use is prohibited.

Introduction - Distribution

I will try to make WinOrbit updates available via some popular
shareware/freeware archives, particularly anonymous ftp from :

ftp.amsat.org : /amsat/software/windows/tracking/winorb28.zip
oak.oakland.edu : /pub3/hamradio/pc/satellite/winorb28.zip
ftp.seds.org : /pub/software/pc/sat/winorb28.zip

As of March 1996, WinOrbit also has a "WWW homepage", where the latest
updates (prior to general release, and not supported) and related
programs such as DDE clients for hardware control can be downloaded.
Thanks to Jean-Phillipe Donnio, (author of The Satellite's Encyclopedia)
and SAT-NET for the space. The URL is :

http://www.sat-net.com/winorbit

The URL checks out as valid.

Cheers,

Rob

8. Re: [HFSIG:2980] Re: Satelite software

From: malberts@ncs.com.na (Mike Alberts)
Date: Mon, 07 Dec 1998 09:24:44 +0200
To: hfsig@tapr.org
Message-ID: <366B82BB.CDFADCF0@ncs.com.na>
Rendered body type: text/plain

Rob Glassey wrote:

> Mike Alberts wrote:
> > > Try WinOrbit:
> > >
> > > ftp://ftp.funet.fi/pub/ham/satellite/tracking/winorb35.zip
> > >
> > > 463 Kbytes
> > Hi Rob,
> > I have errors downloading this file....give me a website address to go
> > to....
>

> WinOrbit is by C.D. Gregory, K8CG.
>
> ------------------------- From the help -------------------------
> Introduction
>
> WinOrbit is a program for computing artificial Earth-satellite position
> and visibility, with the Amateur Radio Satellite operator in mind.
>
> WinOrbit is free. If you enjoy the program and want to express your
> support, you could join AMSAT, or make a donation to the Amateur
> Satellite program, but that's up to you.
>
> Please note: absolutely no guarantees of accuracy or precision are
> made!
> Commercial use is prohibited.
>
> Introduction - Distribution
>
> I will try to make WinOrbit updates available via some popular
> shareware/freeware archives, particularly anonymous ftp from :
>
> ftp.amsat.org : /amsat/software/windows/tracking/winorb28.zip
> oak.oakland.edu : /pub3/hamradio/pc/satellite/winorb28.zip
> ftp.seds.org : /pub/software/pc/sat/winorb28.zip
>
> As of March 1996, WinOrbit also has a "WWW homepage", where the latest
> updates (prior to general release, and not supported) and related
> programs such as DDE clients for hardware control can be downloaded.
> Thanks to Jean-Phillipe Donnio, (author of The Satellite's Encyclopedia)
> and SAT-NET for the space. The URL is :
>
> http://www.sat-net.com/winorbit
>
> --
>
> The URL checks out as valid.
>
> Cheers,
>
> Rob

Hi Rob

The url 's keep telling me no response from servers.......will keep on
trying anyhow to download the software..

Regards
Mike

9. New modems for EVM56002

From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
Date: Mon, 07 Dec 1998 11:28:18 +0100
To: hfsig@tapr.org, dsp@tapr.org
Message-ID: <3.0.5.32.19981207112818.00840200@bicc00.bi.ehu.es>
Rendered body type: text/plain

You can try these new modems for the Motorola EVM56002DSP for AmFax, FmFax
and SSTV, they are intented to be used with JvComm32 program available at:

http://www.jvcomm.de/english/roote.htm

They have been designed by Ulrich Bangert DF6JB for his EasyDSP Unit and
modified by Jabi EA2ARU for use with the EVM.

mailto:govier02@sarenet.es

The files are here:

ftp://det.bi.ehu.es/pub/ham/evm/Fuentes/Amfax95.zip
ftp://det.bi.ehu.es/pub/ham/evm/Fuentes/Fmfax95.zip
ftp://det.bi.ehu.es/pub/ham/evm/Fuentes/Sstv95.zip

        I haven't tried myself, as I don't have the program.

        Eduardo/EA2BAJ

Eduardo Jacob - Area de Ingenieria Telematica
Departamento de Electronica y Telecomunicaciones
ETSII y de IT   Tel: +34-(9)4-427 8055

```
UPV / EHU                   Fax: +34-(9)4-441 4041
Alda Urquijo s/n         E-mail: jtpjatae@bi.ehu.es
E-48013 - Bilbao (Spain)
PGP Key available: Send email w/subject: PGPKEY
fingerprint      : D8 FA 54 49 F7 40 BA AA 09 1A 73 40 A4 26 ED BD
```

10. Re: [HFSIG:2982] New modems for EVM56002

From: Robert McGwier <rwmcgwier@home.com>
Date: Mon, 07 Dec 1998 08:30:27 -0500
To: hfsig@tapr.org, dsp@tapr.org
Message-ID: <366BD872.BA0580FA@home.com>
Rendered body type: text/plain

```
How do I upload these to the TAPR servers?  It took me almost 15 minutes
to get the one window of IP data these files consumed from these
servers.
Eduardo Jacob wrote:
>
> You can try these new modems for the Motorola EVM56002DSP for AmFax, FmFax
> and SSTV, they are intented to be used with JvComm32 program available at:
>
> http://www.jvcomm.de/english/roote.htm
>
> They have been designed by Ulrich Bangert DF6JB for his EasyDSP Unit and
> modified by Jabi EA2ARU for use with the EVM.
>
> mailto:govier02@sarenet.es
>
> The files are here:
>
> ftp://det.bi.ehu.es/pub/ham/evm/Fuentes/Amfax95.zip
> ftp://det.bi.ehu.es/pub/ham/evm/Fuentes/Fmfax95.zip
> ftp://det.bi.ehu.es/pub/ham/evm/Fuentes/Sstv95.zip
>
>         I haven't tried myself, as I don't have the program.
>
>         Eduardo/EA2BAJ
> ----------------------------------------------------------------------
> Eduardo Jacob - Area de Ingenieria Telematica
> Departamento de Electronica y Telecomunicaciones
> ETSII y de IT               Tel: +34-(9)4-427 8055
> UPV / EHU                   Fax: +34-(9)4-441 4041
> Alda Urquijo s/n         E-mail: jtpjatae@bi.ehu.es
> E-48013 - Bilbao (Spain)
> PGP Key available: Send email w/subject: PGPKEY
> fingerprint      : D8 FA 54 49 F7 40 BA AA 09 1A 73 40 A4 26 ED BD
```

11. Re: [HFSIG:2983] Re: New modems for EVM56002

From: Eduardo Jacob <jtpjatae@bicc00.bi.ehu.es>
Date: Mon, 07 Dec 1998 15:24:05 +0100
To: hfsig@tapr.org
Message-ID: <3.0.5.32.19981207152405.008a1470@bicc00.bi.ehu.es>
Rendered body type: text/plain

```
At 07:35 7/12/98 -0600, you wrote:
>How do I upload these to the TAPR servers?  It took me almost 15 minutes
>to get the one window of IP data these files consumed from these
>servers.

        I really don't know what are you talking about. When writing these message
I have seen that only two machines have got these files and they have spend
both of them less than a minute to get each of them.

        Regards

        Eduardo
-----------------------------------------------------
Eduardo EA2BAJ
DSP and HAM:   http://det.bi.ehu.es/~jtpjatae/ham.html
VHF PACKET: EA2BAJ @ EA2URV
PGP Key avlb: Send email w/subject: PGPKEY
fingerprint : D8 FA 54 49 F7 40 BA AA 09 1A 73 40 A4 26 ED BD
```

12. New Modems for EVM56002

From: murrayg@indcom.co.nz
Date: Wed, 9 Dec 1998 7:52:54 +1200
To: hfsig@tapr.org
Message-ID: <199812081852.MAA21769@tapr.org>

Rendered body type: text/plain

I agree with you Eduardo. No trouble downloading your site here.

I think the problem is that folk in the US have trouble with internet
connections outside the US, and particularly with Europe. I experienced
that when I was living there. Probably to do with the American attitude
that they ARE the world!

If there are more problems, folks, let me know, as I have the files and can
easily put them on my site, which is in California.

Murray ZL1BPU

--- snip ---

..
..

Industrial Communications Service Ltd
2/343 Church Street
P.O.Box 12-645
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Auckland
NEW ZEALAND

Tel: +64-9-636-4444
Fax: +64-9-636-4499
www.indcom.co.nz

****
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letterhead, signed by authorised personnel.

****

13. RE: [HFSIG:2985] New Modems for EVM56002

From: Mastenbrook Gary-G13216 <g13216@email1.wes.mot.com>
Date: Tue, 8 Dec 1998 13:54:32 -0600
To: "'hfsig@tapr.org'" <hfsig@tapr.org>
Message-ID: <51F347B016ADD011963200805FC145620340E2D3@email1.wes.mot.com>
Rendered body type: text/plain

No problems reading them from here in Chicago...
73's
Gary N8DMT

Gary Mastenbrook, Technology Manager, Motorola Telematics
50 E. Commerce Drive, Schaumburg, IL 60173
Email: g13216@email.mot.com
Work Home Page: http://www.mot.com
Personal Home Page: http://www.qsl.net/n8dmt

-----Original Message-----
From: murrayg@indcom.co.nz [mailto:murrayg@indcom.co.nz]
Sent: Tuesday, December 08, 1998 12:54 PM
To: hfsig@tapr.org
Subject: [HFSIG:2985] New Modems for EVM56002

I agree with you Eduardo. No trouble downloading your site here.

I think the problem is that folk in the US have trouble with internet
connections outside the US, and particularly with Europe. I experienced
that when I was living there. Probably to do with the American attitude
that they ARE the world!

If there are more problems, folks, let me know, as I have the files and can
easily put them on my site, which is in California.

Murray ZL1BPU

--- snip ---

..
..

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Tel: +64-9-636-4444
Fax: +64-9-636-4499
www.indcom.co.nz

****
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  and may be legally privileged. It does not necessarily represent
  the views nor the official communication of Industrial
  Communications Service Ltd. It is intended solely for the addressee.
  Access to this electronic mail message by anyone else is unauthorised.

  If you are not the intended recipient, disclosure, copying, distribution
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  prohibited and may be unlawful.

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  Service Ltd, will be sent by non-electronic means, on appropriate
  letterhead, signed by authorised personnel.

****

14. Re: [HFSIG:2985] New Modems for EVM56002

From: Bob Johnson <bobj@cisi.com>
Date: Wed, 09 Dec 1998 01:18:02 -0500
To: hfsig@tapr.org
Message-ID: <3.0.2.32.19981209011802.02e71434@mail.atlantic.net>
Rendered body type: text/plain

At 12:54 PM 12/8/98 -0600, you wrote:
>I think the problem is that folk in the US have trouble with internet
>connections outside the US, and particularly with Europe. I experienced
>that when I was living there. Probably to do with the American attitude
>that they ARE the world!

So, when you were living here, did you have the attitude that YOU were the
world? That would explain why you had problems.

Internet connections between the U.S. and some other parts of the world
tend to be very heavily loaded.

-- Bob Johnson
   WB4JCM

15. Doing DSP with LINUX

From: "Pawel Jalocha (as System Administrator of XPCDEL13)" <jalocha@hpdel1.ifj.edu.pl>
Date: Mon, 14 Dec 1998 13:20:39 +0100
To: hfsig@tapr.org
Message-ID: <36750297.D4E95D47@hpdel1.ifj.edu.pl>
Rendered body type: text/plain

Hello to all,

For some time already I have been playing with LINUX and its sound
capabilities.
I find its programming interface to read in the audio quite easy and
universal, which is
not the case for DOS or Windows.

First, to test the DSP capabilities of my PC running under LINUX I wrote

a
decoder for the FSK packet (a la my old PKTMON).
I used C++ as the programming language trying to make objects out
of elementary DSP components like FIR filter, I/Q splitter,
matched tone filter, bit synchronizer, etc.

As a result the full FSK packet receiver (up to the dump of the AX.25
frames)
when running would take some 5% of the CPU. The demodulator was based
on two filters matched to the FSK tones (thus an asynchronous one).
I have then added a second decoding route with a synchronous (phase)
detector and both decoders running at same time would take
less than 10% of my CPU. Once your DSP elements are objects
it is very easy to process same input in several ways at the same time.

My PC is Pentium 133 MHz, 16 MB of RAM and the SoundMan-16 (ISA)
audio card (same as the Pro Audio Spectrum 16-bit).
All programming was done in pure C++, not a single piece in assembler.

Then I have written an FFT routine (yet another FFT..) and measured it
to do
a 256-point complex transform in 600 us (this includes data reordering).

I tried the Phil's one too and it would do the transform in about 250 us

(no data reordering).
To test the FFT I have made a SETI-like spectral "waterflow" display
doing 2048 point real FFT (I packed two real FFTs into one complex).
The graphics was done with the svgalib. After all, using the
1024x768x256
screen resolution, sampling at 7200 Hz I could see the spectra in real
time
with 3.5 Hz resolution.
I could not measure how much CPU it takes because with the svgalibs when
you
switch to another session it holds the graphics execution (?).

But I could make a little SETI run using my TV antenna :-)
You may not believe me, but I could find ETIs almost immediately !
These guys who thing that ETIs are transmitting at 1400 MHz
must be off by a round GHz ! Look around 400 MHz and you will find many:

weak or strong, some with virtually no drift and some moving at 100
Hz/second !
You may not hear them but you can clearly see them on the screen.
Around 400.001 MHz there was a weak carrier moving slowly up and down
by a total of 100 Hz with a period of 5-10 minutes - can their planet
rotate so fast ?

Another good frequency was 150 MHz and there I saw the most astonishing
thing ever:
one of the ETIs was trasmitting... PICOLLO ! I couldn't almost hear it
but on the
spectral display I could clearly see the sequence of tones (being seen
as bright dots)
which were forming a repetitive pattern with a period of 20-30 seconds.
I must write a picollo software now (with LINUX of course),
so I can trasmit a reply up into the skies :-)

Pawel

16. Re: [HFSIG:2988] Doing DSP with LINUX

From: malberts@ncs.com.na (Mike Alberts)
Date: Mon, 14 Dec 1998 14:55:08 +0200
To: hfsig@tapr.org
Message-ID: <36750AAC.7AAA7A7@ncs.com.na>
Rendered body type: text/plain

"Pawel Jalocha (as System Administrator of XPCDEL13)" wrote:

> Hello to all,
>
> For some time already I have been playing with LINUX and its sound
> capabilities.
> I find its programming interface to read in the audio quite easy and
> universal, which is
> not the case for DOS or Windows.
>
> First, to test the DSP capabilities of my PC running under LINUX I wrote

> a
> decoder for the FSK packet (a la my old PKTMON).
> I used C++ as the programming language trying to make objects out
> of elementary DSP components like FIR filter, I/Q splitter,
> matched tone filter, bit synchronizer, etc.
>
> As a result the full FSK packet receiver (up to the dump of the AX.25
> frames)
> when running would take some 5% of the CPU. The demodulator was based
> on two filters matched to the FSK tones (thus an asynchronous one).
> I have then added a second decoding route with a synchronous (phase)
> detector and both decoders running at same time would take
> less than 10% of my CPU. Once your DSP elements are objects
> it is very easy to process same input in several ways at the same time.
>
> My PC is Pentium 133 MHz, 16 MB of RAM and the SoundMan-16 (ISA)
> audio card (same as the Pro Audio Spectrum 16-bit).
> All programming was done in pure C++, not a single piece in assembler.
>
> Then I have written an FFT routine (yet another FFT..) and measured it
> to do
> a 256-point complex transform in 600 us (this includes data reordering).
>
> I tried the Phil's one too and it would do the transform in about 250 us
>
> (no data reordering).
> To test the FFT I have made a SETI-like spectral "waterflow" display
> doing 2048 point real FFT (I packed two real FFTs into one complex).
> The graphics was done with the svgalib. After all, using the
> 1024x768x256
> screen resolution, sampling at 7200 Hz I could see the spectra in real
> time
> with 3.5 Hz resolution.
> I could not measure how much CPU it takes because with the svgalibs when
> you
> switch to another session it holds the graphics execution (?).
>
> But I could make a little SETI run using my TV antenna :-)
> You may not believe me, but I could find ETIs almost immediately !
> These guys who thing that ETIs are transmitting at 1400 MHz
> must be off by a round GHz ! Look around 400 MHz and you will find many:
>
> weak or strong, some with virtually no drift and some moving at 100
> Hz/second !
> You may not hear them but you can clearly see them on the screen.
> Around 400.001 MHz there was a weak carrier moving slowly up and down
> by a total of 100 Hz with a period of 5-10 minutes - can their planet
> rotate so fast ?
>
> Another good frequency was 150 MHz and there I saw the most astonishing
> thing ever:
> one of the ETIs was trasmitting... PICOLLO ! I couldn't almost hear it
> but on the
> spectral display I could clearly see the sequence of tones (being seen
> as bright dots)
> which were forming a repetitive pattern with a period of 20-30 seconds.
> I must write a picollo software now (with LINUX of course),
> so I can trasmit a reply up into the skies :-)
>
> Pawel

Hi All.

I need to get in touch with any HAM that want to help me setting up a HF VHF
gateway from Namibia...

73's
Mike

17. Re: [HFSIG:2988] Doing DSP with LINUX

From: Alberto di Bene <dibene@raleigh.ibm.com>
Date: Mon, 14 Dec 1998 18:45:08 +0100
To: hfsig@tapr.org
Message-ID: <36754EA4.EB5C23F9@usa.net>
Rendered body type: text/plain

Pawel Jalocha (as System Administrator of XPCDEL13) wrote:

> Hello to all,

>
> For some time already I have been playing with LINUX and its sound
> capabilities.
> I find its programming interface to read in the audio quite easy and
> universal, which is
> not the case for DOS or Windows.
> ....

Pawell,
                  myself and Vittorio, IK2CZL, are considering the porting of our pgm
Hamview to Linux, but we do not have enough information on how to handle the audio card,
especially when simultaneous input/output is needed.

Could you (or someone else) please point us to some source of information, preferrably
with code snippets as examples. I mean something for Linux similar to what Ethan Brodsky
has published on Internet for the DOS/Windows world (
http://www.xraylith.wisc.edu/~ebrodsky/)

Thanks and 73
Alberto di Bene, I2PHD

18. Re: [HFSIG:2990] Re: Doing DSP with LINUX

From: W6RCecilA <w6rca@ibm.net>
Date: Mon, 14 Dec 1998 11:54:36 -0700
To: hfsig@tapr.org
Message-ID: <36755EEC.2087@ibm.net>
Rendered body type: text/plain

Alberto di Bene wrote:
> Could you (or someone else) please point us to some source of information, preferrably
> with code snippets as examples. I mean something for Linux similar to what Ethan Brodsky
> has published on Internet for the DOS/Windows world (
> http://www.xraylith.wisc.edu/~ebrodsky/)

I have a similar request for SB-16 compatible PC information. What I need
is the port addresses and setup information or source code for assembly, C,
or BASIC. Most PCs are "SB-16" compatible and I need to know the details.

I would prefer to write in Professional BASIC 7.10 and will make all my
soundboard interface software routines available to everyone via my web page.

Specifically, how does one set up an SB-16 compatible soundcard for 8k
samples per second?
--
73, Cecil, W6RCA   http://people.delphi.com/CecilMoore

19. Re: [HFSIG:2990] Re: Doing DSP with LINUX

From: forrerj@peak.org (Johan Forrer)
Date: Mon, 14 Dec 1998 19:31:22 -0800
To: hfsig@tapr.org
Message-ID: <199812150335.TAA13284@peak.org>
Rendered body type: text/plain

Hi all,

A few pointers that may be of interest.

1) Check out how Hansi uses a sound card under Linux to do PSK31.
     http://wwwcip.informatik.uni-erlangen.de/user/hsreiser/hamradio/

2) Check out Hannu Solainen's work, some commercial, some free, at 4Front
Technologies.
    You will find some pointers to the Linux sound system programming docs.
     http://www.opensound.com/

3) For all sorts of sound editors, FFT scopes, Spectrograms and other fun stuff,
    check out the Linux MIDI & Sound Applications. This is an extensive
collection:
    http://sound.lovebead.com/

4) Lat but not least; check out Tom Sailer's contributions to the 2.2.x
Linux kernel.
    It is a rich collection of DSP demodulators, protocol engines, sound
card drivers,
   and some neat applications running under for Linux. As far as I know that
is available from
    ftp://nic.funet.fi under Linux, but may be at Sunsite or other mirrors
    that carries Linux kernel sources. If you don't know how to find these,
    http://www.linux.org and follow the links. That might be somewhat

owerwhelming
    I fear.

Not all URL's might be 100% recent, so your milage may vary. Make searching
for the
goods part of the discovery process.

If you know of further resources for Linux, be sure to spread the word.

Have fun, 73

--Johan, KC7WW

P.S. Pawel, it's nice to hear from you again.

20. Re: [HFSIG:2990] Re: Doing DSP with LINUX

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Tue, 15 Dec 1998 09:21:37 +0100
To: hfsig@tapr.org
Message-ID: <36761C11.F3971A5B@ife.ee.ethz.ch>
Rendered body type: text/plain

> Could you (or someone else) please point us to some source of information, preferrably
> with code snippets as examples. I mean something for Linux similar to what Ethan Brodsky
> has published on Internet for the DOS/Windows world (
> http://www.xraylith.wisc.edu/~ebrodsky/)

http://www.4front-tech.com has API documentation.

Its basically very simple:
first you open("/dev/dsp", O_RDWR)
then issue a few ioctl's to set sampling parameters,
and subsequently use read/write to input/output samples.
And you can use select/poll to wait for input/output.

Tom

21. Re: [[HFSIG:2992] Re: Doing DSP with LINUX]

From: Alberto di Bene <dibene@usa.net>
Date: 15 Dec 98 15:24:15 MET
To: hfsig@tapr.org
Message-ID: <19981215142416.20411.qmail@www0x.netaddress.usa.net>
Rendered body type: text/plain

Pawel, Tom and Johan,

 thanks for your help. I have added your links to my (ever growing) collection
of useful sites. I had imagined that handling the audio card under Linux were
more complicated. Actually it seems to be much more easier than under DOS (but
the difficulties will show up at coding and testing time, I know.... :-)

Alberto, I2PHD

_____
Get free e-mail and a permanent address at http://www.netaddress.com/?N=1

22. Re: [HFSIG:2993] Re: Doing DSP with LINUX

From: W6RCecilA <w6rca@ibm.net>
Date: Tue, 15 Dec 1998 07:35:26 -0700
To: hfsig@tapr.org
Message-ID: <367673AE.27BE@ibm.net>
Rendered body type: text/plain

Thomas Sailer wrote:
> Its basically very simple:
> first you open("/dev/dsp", O_RDWR)
> then issue a few ioctl's to set sampling parameters,
> and subsequently use read/write to input/output samples.
> And you can use select/poll to wait for input/output.

Dumb question: Does SB16 compatibility imply a DSP microprocessor
on the sound card?
--
73, Cecil, W6RCA   http://people.delphi.com/CecilMoore

23. Re: [[HFSIG:2995] Re: Doing DSP with LINUX]

From: Alberto di Bene <dibene@usa.net>
Date: 15 Dec 98 16:58:59 MET

To: hfsig@tapr.org
Message-ID: <19981215155859.18749.qmail@www0h.netaddress.usa.net>
Rendered body type: text/plain

>Dumb question: Does SB16 compatibility imply a DSP microprocessor
>on the sound card?

Yes, but beware, for the Sound Blaster family of cards, DSP means "Digital
Sound Processor" and not "Digital Signal Processor". It is nothing more than a
fancy microcontroller that handles the various analog switches, volume
controls, etc.
No way to perform with it any kind of real signal processing.

73,
Alberto, I2PHD

_____
Get free e-mail and a permanent address at http://www.netaddress.com/?N=1

24. Re: [HFSIG:2995] Re: Doing DSP with LINUX

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Wed, 16 Dec 1998 10:37:20 +0100
To: hfsig@tapr.org
Message-ID: <36777F50.18B6B106@ife.ee.ethz.ch>
Rendered body type: text/plain

W6RCecilA wrote:

> Dumb question: Does SB16 compatibility imply a DSP microprocessor
> on the sound card?

No. "DSP" in the context of Soundblasters means
"Digital Sound Processor" and is actually only a bad codec.
It cannot do any calculation operations (such as filters)
on the digital sample stream.

Tom

25. Re: [HFSIG:2997] Re: Doing DSP with LINUX

From: W6RCecilA <w6rca@ibm.net>
Date: Wed, 16 Dec 1998 07:00:59 -0700
To: hfsig@tapr.org
Message-ID: <3677BD1B.5F9D@ibm.net>
Rendered body type: text/plain

Thomas Sailer wrote:
>
> W6RCecilA wrote:
>
> > Dumb question: Does SB16 compatibility imply a DSP microprocessor
> > on the sound card?
>
> No. "DSP" in the context of Soundblasters means
> "Digital Sound Processor" and is actually only a bad codec.
> It cannot do any calculation operations (such as filters)
> on the digital sample stream.

Thanks Tom. When my laptop says "SB16 compatible" does that
mean hardware compatible or driver compatible. Thanks.
--
73, Cecil, W6RCA http://people.delphi.com/CecilMoore

26. Re: [HFSIG:2996] Re: [Re: Doing DSP with LINUX]

From: W6RCecilA <w6rca@ibm.net>
Date: Wed, 16 Dec 1998 07:02:21 -0700
To: hfsig@tapr.org
Message-ID: <3677BD6D.35B6@ibm.net>
Rendered body type: text/plain

Alberto di Bene wrote:
>
> >Dumb question: Does SB16 compatibility imply a DSP microprocessor
> >on the sound card?
>
> Yes, but beware, for the Sound Blaster family of cards, DSP means "Digital
> Sound Processor" and not "Digital Signal Processor". It is nothing more than a
> fancy microcontroller that handles the various analog switches, volume
> controls, etc.

> No way to perform with it any kind of real signal processing.

Thanks Alberto, when my laptop says "SB16 compatible" does that mean
hardware compatible or driver compatible? Thanks.
--
73, Cecil, W6RCA http://people.delphi.com/CecilMoore

27. Re: [HFSIG:2999] Re: [Re: Doing DSP with LINUX]

From: Alberto di Bene <dibene@raleigh.ibm.com>
Date: Wed, 16 Dec 1998 16:08:27 +0100
To: hfsig@tapr.org
Message-ID: <3677CCEB.3D86686F@usa.net>
Rendered body type: text/plain

W6RCecilA wrote:
>....
>Thanks Alberto, when my laptop says "SB16 compatible" does that mean
>hardware compatible or driver compatible? Thanks.
>....

Generally the laptops aren't really hardware compatibles with the SB16. I still have to
find one which is (this doesn't of course mean it doesn't exist). The 16-bit WAV
compatibility is achieved using an ad-hoc driver for the specific chip used in the
laptop. This precludes the use of programs that talk directly to the hardware.

73, Alberto, I2PHD

28. Re: [HFSIG:3000] Re: [Re: Doing DSP with LINUX]

From: Doug Faunt N6TQS +1-510-655-8604 <faunt@netcom.com>
Date: Wed, 16 Dec 1998 09:19:00 -0800 (PST)
To: hfsig@tapr.org
Cc: hfsig@tapr.org
Message-ID: <199812161719.JAA24262@netcom4.netcom.com>
Rendered body type: text/plain

Date: Wed, 16 Dec 1998 09:11:48 -0600 (CST)
   From: Alberto di Bene <dibene@raleigh.ibm.com>

   W6RCecilA wrote:
   >....
   >Thanks Alberto, when my laptop says "SB16 compatible" does that mean
   >hardware compatible or driver compatible? Thanks.
   >....

   Generally the laptops aren't really hardware compatibles with the SB16. I still have to
   find one which is (this doesn't of course mean it doesn't exist). The 16-bit WAV
   compatibility is achieved using an ad-hoc driver for the specific chip used in the
   laptop. This precludes the use of programs that talk directly to the hardware.

   73, Alberto, I2PHD

One possible exception: W6OTC is reported to have run RITTY by K6STI
on a Micron laptop with the Vibra 16 soundcard.
Check N1RCT's RTTY DXing page, under "perfect dxpedtioning rig" for
the reference.
73, doug

29. Re: [HFSIG:3001] Re: [Re: Doing DSP with LINUX]

From: Thomas Sailer <sailer@ife.ee.ethz.ch>
Date: Thu, 17 Dec 1998 09:40:56 +0100
To: hfsig@tapr.org
Message-ID: <3678C398.67C0FC11@ife.ee.ethz.ch>
Rendered body type: text/plain

Doug Faunt N6TQS +1-510-655-8604 wrote:

> One possible exception: W6OTC is reported to have run RITTY by K6STI
> on a Micron laptop with the Vibra 16 soundcard.

This is no exception because Vibra 16 actually is a soundblaster 16.
The "SB16" Chip is usually called Vibra 16.

Tom

30. AMBE1000+ RFC

From: Charles Brain <chbrain@dircon.co.uk>
Date: Fri, 25 Dec 1998 17:11:53 +0000
To: hfsig@tapr.org, ss@tapr.org

Message-ID: <3683C759.9C160AB4@dircon.co.uk>
Rendered body type: text/plain

Hello Everyone,

In furtherance of my digital speech experiments, I am now laying out a
digital speech board. This board is based around the AMBE1000+ low rate Vocoder
chip.

For further details see

http://www.dvsinc.com/

I have chosen to go the chip approach mainly due to licensing concerns
over the MELP algorithm that I have been experimenting with. Plus the MELP
approach needs lots of MIPS whereas the AMBE draws 65ma max and the chip is
only one inch square.

Originally I was only going to use the board to interface to a Motorola EVM
running a parallel tone modem I am working on. However the board will have
a processor on it so would be ideal for other applications.

This is where you can help!

My current design consists of the following

Motorola MC14LC5480	Codec
AMBE-1000	Vocoder
PIC16C67	CPU (maybe)
MAX232	RS232 Level convertor
74HC04	Interface logic codec <-> vocoder
74HC74	" " " "
74HC4060	Clock 2.048Mhz and 8Khz
7805	Voltage Regulator.
3 Xtals	as I can't find an easy way to only use one!

The microphone input will be balanced with a supply for an electret unit.
I think the codec is designed to interface with a telephone handset. I have added
a PTT line as well, although the chip has VAD, it also supports DTMF tone generation and
can provide comfort noise and echo cancellation (the echo cancellation is for 2 to 4 wire
hybrids and is of little use to me).

The board will power up at 2400 bps speech with no FEC, although it can be reprogrammed
via the PIC to any rate in 50 baud increments up to 9600 bps with different rates of FEC.

All the components except the Vocoder are DIP packages, the vocoder only
comes in a TQFP package. As I am configuring the chip via the PIC most of the
pins on the vocoder are either at 5v or grounded. I can't finnish the board off
until I get the TQFP packages as the PCB software does not come with a TQFP
footprint so I have had to make one and I flunked technical drawing at school
so I want to check it against an actual component!

I could have done the whole thing using SMT but it is easier to work on this way,
I don't know what people feel about this?

I have used quite a large PIC, mainly for two reasons, firstly to provide space
for future software expansion and secondly, quite a few people seem to know how
to program these chips. Hopefully someone will feel inspired into making a digital
2m radio! I think all the call control software would have to run on a separate
CPU though. It should be possible to do a radio that can do full duplex on a single
simplex channel using a Ping-Pong type protocol.

The actual board will be about 1/2 the size of the Motorola EVM. I hope to have
the first unit ready by the end of January. The vocoders should arrive the first week
of January as DVS have received the money I wired them and the chips are in stock. All the
other parts are available here in the U.K.

I have added an RS232 interface so it can be connected to the Host port of an EVM.

The parts cost should work out to about $150.

I don't want to make the board too complicated, but if anyone has any suggestions as to things
that would make it more useful to them, please let me know. So far I have laid out the Codec,
interface logic, psu parts and the RS232 interface. I have not finally decided on the CPU yet.
But I will probably use a PIC unless someone has a better idea.

The end of January time frame is for the first prototype board, the software will take a bit longer!

- Charles G4GUO

31. New versions of PSK31 by G3PLX: 56002EVM on Windows and SB on Windows

From: Eduardo Jacob <edu@kender.es>
Date: Sat, 26 Dec 1998 09:00:31 -0100
To: psk31@bipt106.bi.ehu.es
Cc: hfsig@tapr.org, dsp@tapr.org
Message-ID: <3.0.5.32.19981226090031.0085f570@kender.es>
Rendered body type: text/plain

As Christmas time is allways full of surprises and gifts Peter has produced
two new versions of PSK31:

For Motorola EVM56002 on Windows available at:
ftp://det.bi.ehu.es/pub/ham/psk31/psk31evm.zip

Another for SoundBlaster (Yes, SoundBlaster and hopefully compatibles) on
Windows, available at:

ftp://det.bi.ehu.es/pub/ham/psk31/psk31sbw.zip

These new versions have new tunning and level mettering helps.

Regards

Eduardo

32. PSK31

From: Brian Beezley <k6sti@n2.net>
Date: Sat, 26 Dec 1998 07:16:07 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19981226071607.19af021a@mail.n2.net>
Rendered body type: text/plain

>As Christmas time is allways full of surprises and gifts Peter has produced
>two new versions of PSK31:

Terrific! I don't know what the PSK31 activity frequencies are but I'll be
CQing this weekend with PSK31 on 14080 and 7080.

Brian

33. PSK31 Calling Freq

From: Brian Beezley <k6sti@n2.net>
Date: Sat, 26 Dec 1998 10:28:04 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19981226102805.1a17e262@mail.n2.net>
Rendered body type: text/plain

G3PLX and KC7WW tell me that 14070.15 kHz is the activity frequency for
PSK31 on 20 meters. I'll be CQing there during the daytime and on 7070.15
kHz at night this weekend. Earlier today I worked KC7WW on a random PSK31
CQ. Fun!

To repeat Eduardo's announcement, you can download PSK31 for Win 3.1/95/98 at

ftp://det.bi.ehu.es/pub/ham/psk31/psk31sbw.zip

The program is easy to get going and simple to operate. Since my computer
was already cabled for RTTY using a sound card I didn't have to touch a
thing. It just came up and ran. Hope to see others on PSK31 soon.

Brian

34. Re: [HFSIG:3003] AMBE1000+ RFC

From: "Louis Cobet" <louis_cobet@email.msn.com>
Date: Sat, 26 Dec 1998 14:28:19 -0800
To: <hfsig@tapr.org>
Message-ID: <000801be311f$0ec21460$271c2599@hp-customer>
Rendered body type: text/plain

You wrote:

>In furtherance of my digital speech experiments, I am now laying out a
>digital speech board. This board is based around the AMBE1000+ low rate
Vocoder
>chip.
>
>For further details see   ....

I am interested in this project. I don't understand why more has not been
done in digital voice by hams.

RF Micro Devices, http://www.rfmd.com, has a number of digital transceiver
chips that would be useful.

9.6 K would permit FEC. One of the more interesting questions is what kind
of transmission protocol to use. We need a standard for amateur digital
voice.

Louis Cobet, K6MDH
louis_cobet@email.msn.com

35. Re: [HFSIG:3007] Re: AMBE1000+ RFC

From: Charles Brain <chbrain@dircon.co.uk>
Date: Sun, 27 Dec 1998 10:27:39 +0000
To: hfsig@tapr.org
Message-ID: <36860B9B.7C2EF0E0@dircon.co.uk>
Rendered body type: text/plain

Louis Cobet wrote:

> I am interested in this project. I don't understand why more has not been
> done in digital voice by hams.
>

> 9.6 K would permit FEC. One of the more interesting questions is what kind
> of transmission protocol to use. We need a standard for amateur digital
> voice.
>

Hello Louis,

Well I think you have answered your own question! In a previous existence
I spent a while investigating the implementation of a GSM base station.
The protocol documentation is huge! Also most of the intelligence is in the
network and not the mobile. In a ham radio environment we would probably
want the intelligence in the mobile. That is one reason why I feel the
Amateur service can contribute to the radio art in this field, i.e the design
of an infrastructureless system.

The other problem is expertise and time, most of the people that have
the expertise don't have the time and visa versa.

Also we probably would not be able to agree on whether it should be a SS or TDMA
system :-)

I am doing this board for my own use primarily. Its output is via
RS232 for direct connection to my EVM. But it would be quite possible
to stick a DOS SBC in between the Vocoder and the DSP and implement a
call control protocol on that.

I have a spare JKMicro board here, it has 2 serial ports 20 IO lines, 512K of
memory and a 10 Mhz CPU and runs a flash based DOS, all that for just
over $100, less in quantity. something like that would be good for a
proof of concept system.

Credit where credit is due Doug Braun pointed these chips out to me.
Judging by the DVS Inc press release a lot of the new LEO systems will
be using them, the chips are $48 each in small quantities.

- Charles

36. Re: [HFSIG:3006] PSK31 Calling Freq

From: "Louis Cobet" <louis_cobet@email.msn.com>
Date: Mon, 28 Dec 1998 17:26:37 -0800
To: <hfsig@tapr.org>
Message-ID: <001801be32ca$4a7c5f40$820e2599@hp-customer>

Rendered body type: text/plain

In addition to 14070.15 kHz and 7070.15 kHz as calling frquencies, why not
add a ten meter frequency to the list. Offhand, I suggest 28130.

Louis Cobet, K6MDH
louis_cobet@email.msn.com

37. Re: [HFSIG:3009] Re: PSK31 Calling Freq

From: Brian Beezley <k6sti@n2.net>
Date: Mon, 28 Dec 1998 17:59:07 -0800 (PST)
To: hfsig@tapr.org
Message-ID: <1.5.4.16.19981228175907.1a372aa6@mail.n2.net>
Rendered body type: text/plain

At 07:29 PM 12/28/98 -0600, you wrote:
>In addition to 14070.15 kHz and 7070.15 kHz as calling frquencies, why not
>add a ten meter frequency to the list. Offhand, I suggest 28130.
>
>Louis Cobet, K6MDH
>louis_cobet@email.msn.com
>
>
>

28070.15 currently is active with PSK31. I'm QRV for the 0.15 story.

Brian

38. TAPR FTP Site

From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
Date: Tue, 29 Dec 1998 00:57:57 -0600
To: "HF SIG list   mailing"<hfsig@tapr.org>, " Spread Spectrum "<ss@tapr.org>,
        " TAPR/AMSAT DSP "<dsp@tapr.org>,
        "APRS SIG list   mailing"<aprssig@tapr.org>,
        "NETSIG list   mailing"<netsig@tapr.org>,
        "BBS SIG list   mailing"<bbssig@tapr.org>,
        " tacgps "<tacgps@tapr.org>, aprsnews, mic-e,
        TAPR Regional Freq <regional_freq@tapr.org>
Cc: "Bill Westradio   Pasternak"<billwa6itf@aol.com>,
        "Len KB7LPW   Winkler"<lenwink@indirect.com>,
        "Hap KC9RP RAIN Holly"<hholly@ais.net>,
        "Greg Jones WD5IVD" <wd5ivd@tapr.org>, "Rich   CQ VHF"<CQVHF@aol.com>,
        "Gagne, Jennifer, N1TDY" <jgagne@arrl.org>,
        "Donald   Rotolo"<73227.2644@compuserve.com>,
        "Stan   Horzepa"<stanzepa@nai.net>,
        " Bob Hansen (TAPR) "<psr@tapr.org>,
        "Ham Radio On-line" <vbook@vbook.com>,
        " Tom McDermott "<mcdermot@eagle.aud.alcatel.com>,
        <rlindquist@arrl.org>, "Brad Smith, KC5SP" <brads@galstar.com>
Message-ID: <v04020a20b2ae2874411e@[207.43.172.66]>
Rendered body type: text/plain

Please be aware that the TAPR FTP site (ftp://ftp.tapr.org) has undergone a
major reorganization as part of the upcoming web site restructuring and
redesign.

If you have links on web pages that reference into our ftp site, please
take a moment to check them and update those URLs to reflect the changes at
our site.

The shallower directory structure should make things easier for individuals
visiting our site to find files and information.

Sorry for any inconvenience this might cause.

Cheers - Greg Jones, WD5IVD
TAPR

The new FTP directory structure:

ftp://ftp.tapr.org
Top Level:
        aprssig/
                WinMacData/

```
        aprsnews/
        dosstuff/
                APRSdos/
                DOSmisc/
        garmin/
        history/
        javastuff/
        linux/
        macstuff/
                MacAPRS/
                MacGPS/
                MacMisc/
                PacketTracker/
        mail_archive/
        maps/
                100K_Not_On_CD/
                MacMaps/
                PCMaps/
                WinMacMaps/
        mic-e/
        movies/
        palmstuff/
                palmaprs/
                palmmaps/
                palmmisc/
                palmos/
        presentations/
        shlib/
        upload/
        winstuff/
                APRSPLUS/
                Waypoint/
                WinAPRS/
                WinMisc/

bbssig/
        mail_archive/

das/
        mail_archive/

dcc/

dsp/
        dsp56000/
        dsp56001/
        dsp56002/
        mail_archive/
        psa_sound_card/
        recent_uploads/
        tms320c10/
        tms320c26/
        tms320c50/
        upload/

dsp93/
        Paccomm_eproms/
        diskettes/
        dsp93_eproms/
        mail_archive/
        manuals/
        schems/
        simulators/
        software/
        software.list
        source/
        stlouis.wrkshp.95/
        updates/
        upload/
        util/

general/
        9600baud/
                ALINCO/
                ICOM/
                KENWOOD/
                MOTOROLA/
                YEASU/
```

```
        gps/
                HPAN52-2/
                garmin/
                motorola/

        hfsig/
                mail_archive/

        info/
                kits/
                logos/
                publications/

        netsig/
                mail_archive/

        psr/

        pub/

        regional_freq/
                mail_archive/

        software_lib/
                DEMOS/
                Linux/
                RF/
                UPLOAD/
                an93/
                bbs/
                dsp/
                gps/
                listing
                misc/
                sat/
                switch/
                tcpip/
                terminal/
                thenet.txt
                tnc/
                utils/
                weather/

        ss/
                mail_archive/
                SS_STA/
                dcc97_ss_seminar/

        tac2/
                mail_archive/

        talnet/

        tapr-bb/
                mail_archive/

        tapr-tnc/
                mail_archive/

        utils/

-----
Greg Jones, WD5IVD                 Austin, Texas
wd5ivd@tapr.org      http://www.tapr.org/~wd5ivd
```

39. Updated Web page.

```
From: forrerj@peak.org (Johan Forrer)
Date: Tue, 29 Dec 1998 21:30:59 -0800
To: HFSIG@tapr.org
Message-ID: <199812300534.VAA18612@peak.org>
Rendered body type: text/plain
```

Hi,

Just thought I'd let folks know that I have updated my web page
after a long dry spell. If you were looking some some of the DSP
and radio article test --- that is there now. Also a bit new material
on Meteor activities during the recent Leonids shower -- fun stuff.

I also have cleaned up dead links and made some older code
packages available for downloading.

73

Johan Forrer

http://www.peak.org/~forrerj

40. New SIG for the Kenwood THD7A/E

From: "Horzepa, Stan" <Stan_Horzepa@adc.com>
Date: Wed, 30 Dec 1998 12:57:46 -0600
To: AO16APRS <ao16aprs@tapr.org>, APRS-FL <aprs-fl@tapr.org>,
APRSNEWS
 <aprsnews@tapr.org>, APRSSIG <aprssig@tapr.org>,
BBSSIG
 <bbssig@tapr.org>, DSP sig <dsp@tapr.org>,
HFSIG <hfsig@tapr.org>, HTAPRS <htaprs@tapr.org>,
MIC-E sig <mic-e@tapr.org>, NETSIG
 <netsig@tapr.org>,
PROPNET sig <propnet@tapr.org>,
REGIONAL_FREQ sig
 <regional_freq@tapr.org>, SS sig <ss@tapr.org>,
TACGPS sig
 <tacgps@tapr.org>, TAPR-BB sig <tapr-bb@tapr.org>,
TAPR-TNC
 <tapr-tnc@tapr.org>
Message-ID: <1B66FFC6BE1CD211A0620008C7A42AC21EF760@MRDNEXCH01.ct.adc.com>
Rendered body type: text/plain

TAPR is pleased to announce the formation of a new special
interest group, the HTAPRS SIG.

The purpose of HTAPRS is to provide a forum for the discussion
of issues related to the Kenwood THD7A/E transceiver. (HTAPRS
is also intended to lessen the number of messages on TAPR's
APRSSIG by moving THD7A/E traffic off that SIG.)

To subscribe to the HTAPRS mail list, send email to
listserv@tapr.org with the following text in the body of the
message:

   subscribe htaprs your_first_name your_last_name

41. PSK31 on RTTY list

From: Doug Faunt N6TQS +1-510-655-8604 <faunt@netcom.com>
Date: Wed, 30 Dec 1998 17:42:52 -0800 (PST)
To: hfsig@tapr.org, dsp@tapr.org, bpsk@qth.net
Message-ID: <199812310142.RAA14808@netcom9.netcom.com>
Rendered body type: text/plain

Just recently, there's a lot of interest and discussion of PSK31 on the
wf1b-rtty@wf1b.com RTTY list.
73, doug

42. Re: [HFSIG:3011] HFSIG digest 620

From: Art Sutorus <asutoru@pe.net>
Date: Thu, 31 Dec 1998 05:52:16 -0800
To: hfsig@tapr.org
Message-ID: <368B8190.93EC5EDC@pe.net>
Rendered body type: text/plain

Please UNSUBSCRIBE. Tnx.

hfsig@tapr.org wrote:

> HFSIG Digest 620
>
> Topics covered in this issue include:
>
>   1) Re: PSK31 Calling Freq
>   by Brian Beezley <k6sti@n2.net>
>   2) TAPR FTP Site
>   by "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
>
> --
>
> Date: Mon, 28 Dec 1998 17:59:07 -0800 (PST)
> From: Brian Beezley <k6sti@n2.net>

> To: hfsig@tapr.org
> Subject: Re: PSK31 Calling Freq
> Message-ID: <1.5.4.16.19981228175907.1a372aa6@mail.n2.net>
>
> At 07:29 PM 12/28/98 -0600, you wrote:
> >In addition to 14070.15 kHz and 7070.15 kHz as calling frquencies, why not
> >add a ten meter frequency to the list. Offhand, I suggest 28130.
> >
> >Louis Cobet, K6MDH
> >louis_cobet@email.msn.com
> >
> >
> >
>
> 28070.15 currently is active with PSK31. I'm QRV for the 0.15 story.
>
> Brian
>
> ------------------------------
>
> Date: Tue, 29 Dec 1998 00:57:57 -0600
> From: "Greg Jones, WD5IVD" <wd5ivd@tapr.org>
> To: "HF SIG list   mailing"<hfsig@tapr.org>, " Spread Spectrum "<ss@tapr.org>,
> Cc: "Bill Westradio   Pasternak"<billwa6itf@aol.com>,
> Subject: TAPR FTP Site
> Message-ID: <v04020a20b2ae2874411e@[207.43.172.66]>
>
> Please be aware that the TAPR FTP site (ftp://ftp.tapr.org) has undergone a
> major reorganization as part of the upcoming web site restructuring and
> redesign.
>
> If you have links on web pages that reference into our ftp site, please
> take a moment to check them and update those URLs to reflect the changes at
> our site.
>
> The shallower directory structure should make things easier for individuals
> visiting our site to find files and information.
>
> Sorry for any inconvenience this might cause.
>
> Cheers - Greg Jones, WD5IVD
> TAPR
>
> ----
>
> The new FTP directory structure:
>
> ftp://ftp.tapr.org
> Top Level:
>   aprssig/
>   WinMacData/
>   aprsnews/
>   dosstuff/
>   APRSdos/
>   DOSmisc/
>   garmin/
>   history/
>   javastuff/
>   linux/
>   macstuff/
>   MacAPRS/
>   MacGPS/
>   MacMisc/
>   PacketTracker/
>   mail_archive/
>   maps/
>   100K_Not_On_CD/
>   MacMaps/
>   PCMaps/
>   WinMacMaps/
>   mic-e/
>   movies/
>   palmstuff/
>   palmaprs/
>   palmmaps/
>   palmmisc/
>   palmos/
>   presentations/
>   shlib/

```
>                 upload/
>                 winstuff/
>                         APRSPLUS/
>                         Waypoint/
>                         WinAPRS/
>                         WinMisc/
>
>         bbssig/
>                 mail_archive/
>
>         das/
>                 mail_archive/
>
>         dcc/
>
>         dsp/
>                 dsp56000/
>                 dsp56001/
>                 dsp56002/
>                 mail_archive/
>                 psa_sound_card/
>                 recent_uploads/
>                 tms320c10/
>                 tms320c26/
>                 tms320c50/
>                 upload/
>
>         dsp93/
>                 Paccomm_eproms/
>                 diskettes/
>                 dsp93_eproms/
>                 mail_archive/
>                 manuals/
>                 schems/
>                 simulators/
>                 software/
>                 software.list
>                 source/
>                 stlouis.wrkshp.95/
>                 updates/
>                 upload/
>                 util/
>
>         general/
>                 9600baud/
>                         ALINCO/
>                         ICOM/
>                         KENWOOD/
>                         MOTOROLA/
>                         YEASU/
>
>         gps/
>                 HPAN52-2/
>                 garmin/
>                 motorola/
>
>         hfsig/
>                 mail_archive/
>
>         info/
>                 kits/
>                 logos/
>                 publications/
>
>         netsig/
>                 mail_archive/
>
>         psr/
>
>         pub/
>
>         regional_freq/
>                 mail_archive/
>
>         software_lib/
>                 DEMOS/
>                 Linux/
>                 RF/
>                 UPLOAD/
```

```
>                 an93/
>                 bbs/
>                 dsp/
>                 gps/
>                 listing
>                 misc/
>                 sat/
>                 switch/
>                 tcpip/
>                 terminal/
>                 thenet.txt
>                 tnc/
>                 utils/
>                 weather/
>
>         ss/
>                 mail_archive/
>                 SS_STA/
>                 dcc97_ss_seminar/
>
>         tac2/
>                 mail_archive/
>
>         talnet/
>
>         tapr-bb/
>                 mail_archive/
>
>         tapr-tnc/
>                 mail_archive/
>
>         utils/
>
> -----
> Greg Jones, WD5IVD                 Austin, Texas
> wd5ivd@tapr.org      http://www.tapr.org/~wd5ivd
>
> ------------------------------
>
> End of HFSIG Digest 620
> ************************
```

43. ESDI disk, Quiktrak Y2K, Timewave DSP2232 source

From: Robert McGwier <rwmcgwier@home.com>
Date: Thu, 31 Dec 1998 19:17:09 -0500
To: amsat-bb@amsat.org, dsp@tapr.org, hfsig@tapr.org
Message-ID: <368C1405.316B62F0@home.com>
Rendered body type: text/plain

Howdy:

I have an ESDI drive with Quiktrak source, Timewave (AEA) DSP2232 source, the C compiler used to compile Quiktrak, source code
for many other things of great importance to me, ASM56000 assembler, signals development tools that I wrote and would like to have.

Let's try this again. The help that has been offered to me so far, I believe I can do myself but would NOT like to do, that is, plug the Ultrastor ESDI card into an available ISA slot and then copy off what
is needed one slow floppy at a time. I don't have the cables to connect the ESDI drive to the card. I don't have a tape back up as I keep copies of everything, elsewhere. My ultimate wish list for this would be find someone who already has an ESDI card, ready to go, in some old machine that they can easily get the stuff off and onto to some mass storage device that I can ftp and get the data to my development machine. The entire drive is 150 Meg's and the stuff I want is not so large as this.

If anyone thinks they can and will help, let me know.

Bob